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Eighth Semiannual Technical Report

March 1977

For the Project

INTEGRATED DOD VOICE & DATA NETWORKS
AND GROUND PACKET RADIO TECHNOLOGY

VOLUME 1 - PART 1

INTEGRATED DOD VOICE & DATA NETWORKS

network analysis corporation

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For the Project
INTEGRATED DOD VOICE AND DATA NETWORKS
AND GROUND PACKET RADIO TECHNOLOGY

VOLUME 1 PART 1

INTEGRATED DOD VOICE AND DATA NETWORKS

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EIGHTH SEMIANNUAL TECHNICAL REPORT

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NETWORK ANALYSIS CORPORATION

Eighth Semiannual Technical Report

Integrated DOD Voice and Data Networks and Ground Packet Radio Technology

March 1977

EXECUTIVE SUMMARY

I. PROGRAM ELEMENTS AND OVERALL OBJECTIVES

NAC's current project, for which this Semiannual Report is an interim report, has three major components and sets of objectives:

1. Integrated DOD Voice and Data Networks -

   - The project's specific concern is to identify the appropriate mix of switching technologies (e.g., circuit, packet, and integrated switching) that can best meet DOD data and voice communications requirements in separate or integrated future networks. To perform the analyses necessary for this determination, the project's goals are to identify key issues and parameters, to develop cost/performance tradeoffs and technology assessments, and to provide detailed recommendations for best meeting projected DOD requirements for voice and data communications in the 1980's and beyond.

2. Gateway Topological Optimization for Interconnecting Packet Switched Networks -

- Growth of different packet switching networks within the DOD is leading to a number of questions concerning the most effective means to connect these networks when necessary. A major goal of this project is to develop techniques for optimizing the number and locations of gateways required to interconnect packet networks. Additionally, these techniques will be utilized to identify fundamental parameters influencing gateway locations, and as a concrete example, gateway strategies for connecting ARPANET and AUTODIN II will be recommended.

3. Ground Packet Radio Technology -

ARPA has developed a ground packet radio system that is currently undergoing a sequence of experimental tests and performance verifications. Tests are constrained because of the limited number of repeaters which presently exist and because of the complexity of simulating all possible environments and stresses experimentally. The goals of this project are:

- To determine (via simulation and analysis) the performance profile of packet radio networks as a function of key system parameters such as maximum capacity, error rates, and equipment limitations.

- To provide specific recommendations for the enhancement of the experimental packet radio system in order to improve factors such as the time required to stabilize after element failures and system capacity under noisy conditions.

- To determine the speed and characteristics of devices within a packet radio system whose elements are mobile. To determine the range for which the present network procedures become infeasible or introduce unacceptably high performance degradations, and to propose feasible system design alternatives for use within mobile networks.

II. RESULTS PRESENTED IN THIS REPORT

In this section we present the summary of each chapter included in this Eighth Semiannual (Interim) Technical Report. The report is organized into four volumes. Volume 1 covers the accomplishments in the area of integrated DOD voice and data networks. In Volume 2 we present cost trends for large volume packet data networks, incorporating satellite technology, local access, and switch considerations. Volume 3 presents the research and experiments in the area of topological gateway placement strategies, and Volume 4 is dedicated to the area of packet radio technology. The summary by chapter follows.

CHAPTER 1: A CLASSIFICATION OF ROUTING STRATEGIES FOR TELECOMMUNICATIONS

This chapter presents a taxonomy for classification of telecommunication routing algorithms. The taxonomy developed is an essential ingredient of the integrated DOD voice and data networks study, since with modern communication and processing technologies, the differences between routing in packet switched networks, advanced circuit switched network concepts, message switched networks, etc., are no longer apparent. The taxonomy presented uses more elementary features such as: time, space, message types, as the basis for classification of routing strategies. Apart from enabling the classification of known routing algorithms previously characterized as "adaptive", "deterministic", "centralized", "distributed", etc., it enables us to:

- Understand the processes which comprise a routing strategy, and the interrelationship between the various message types and time parameters of the strategy.

- Synthesize new routing strategies with desired properties.

On the basis of this research, we formulated and implemented two routing algorithms for circuit switched networks as a stand alone switching alternative or as part of an integrated circuit-packet switching network. The routing algorithms are currently being tested.

The utility of classification on the basis of the above elementary features enables specification of processing and storage requirements at switching nodes in support of routing decisions. The following issues can specifically be addressed using this classification:

- Items to be measured (e.g., number of hops, delays)

- Frequency of measuring various quantities

- Processing of measured information

- Destinations and frequency for disseminating measurements.

CHAPTER 2: ANALYSIS OF INTEGRATED SWITCHING LINKS

The analysis of an integrated switching link is addressed in this chapter. The notion of an integrated switching network (of which the link is an element) considered here is one in which traffic can be served on a blocking or delay basis. The integrated switching concept includes elements from circuit switching and elements from packet switching. The switching and transmission facilities of the network are dynamically shared between traffic using the circuit and packet switched modes of operation.

The notion of circuit-switching as addressed in this chapter does not coincide with the classical notion where a physical end-to-end path is dedicated to a pair engaged in communication. The notion considered resembles a reservation approach where the network control programs, rather than the end users, manage and control the end-to-end resources on a real-time basis.

The analytical models address two levels of such integration, the so-called "fixed boundary" in which the link capacity is partitioned between circuit switched and packet switched modes of operation, and the so-called "moveable boundary" where "data", using the packet switched mode, can utilize idle voice capacity.

The models developed have the following properties and capabilities:

- Fixed and moveable boundary frame management.
- Voice and data priorities and precedence.
- Includes parameters of voice digitalization rates and packet size.
- Store-and-forward traffic with different packet sizes for signaling and information.

Quantitative results presented investigate frame partition and frame management policies, sensitivity to total traffic load and traffic mix, speech interpolation, and alternatives for incorporating signaling traffic on an integrated link. These studies indicate that:

- The moveable boundary strategy which enables dynamic sharing of channel capacity between the switching modes is superior in that it minimizes the cost of transmission facilities. For normal operating conditions (1% blocking), using the AUTOVON/AUTODIN II requirements, improvement in channel capacity ranges between 12% and 25%.
- The resolution of fixed vs. moveable boundary is dependent upon the fraction of circuit and packet switched traffic, the voice digitization rates employed, the distribution of message length using the packet switched mode,

and the blocking and average delay values for which the network is engineered.

- If links are engineered for high blocking probability the advantages of a moveable boundary are significantly diminished. Hence the complexity and cost of an integrated switch may not offset the savings in channel capacity.

- A similar conclusion to that above results when the appropriate packet slot size is large compared to the slot size used for circuit switched traffic and no packet fragmentation is allowed. This may be the case when a low voice digitization rate is used for voice in the circuit switched mode.

CHAPTER 3: DESIGN OF INTEGRATED SWITCHING NETWORKS

The design of integrated switching networks, a problem which has not been addressed in the literature to date, is presented in this chapter. The design of integrated networks encompasses all the elements (subproblems) of circuit switched network design and packet switched network design. In addition, it includes functions which evolve from the dynamic sharing of resources by the two modes.

Briefly stated, the problem requires the determination of minimum cost network resources (nodes, links, capacities) which satisfy average end-to-end delay for packet switched traffic, end-to-end loss probability for circuit switched traffic, average end-to-end delay for circuit connection set up, and reliability constraints.

The results of this chapter are divided into two parts. In the first part we present procedures and algorithms and in the second part we discuss computational techniques and describe the

capabilities of an interactive computer program and a sample application to the AUTOVON and AUTODIN II traffic data bases. Results include:

- Performance variables, design variables, and constraints for integrated network design are defined. Two design performance measures for the circuit switched subnet were derived. These relate the user performance measure (the average end-to-end loss probability, P_E) to design performance measures. One performance function relates P_E to the average link blocking probability and the second function relates P_E to the average path blocking probability.

- Routing for circuit switched and packet switched modes of operation which take advantage of the integrated network concept are developed.

- An optimization procedure for determing integrated link capacities under fixed and dynamic sharing of the capacity between circuit and packet switched modes.

- An overall procedure for the total topological design of integrated networks has been implemented and tested. This represents a significant advance in the state of the art.

A detailed presentation of the capabilities of the interactive program developed for integrated network design is given in this chapter. The program engineers the network for voice and data traffic. The major options of the program are:

- Fixed and moveable boundary frame management strategies for integrated links.
- Up to six classes of traffic for the packet switching mode, with possible different end-to-end delay requirements.
- Design can accommodate traffic classes with priorities.
- Signaling traffic for circuit connection set-up and circuit disconnection can be accommodated on dedicated channels or by sharing capacity with packet switched traffic - with or without priorities.

This design procedure will be used extensively to develop cost/ performance tradeoffs during the remainder of the project.

CHAPTER 4: NETWORK MODELS FOR PACKETIZED SPEECH

In this chapter we present basic research results leading to a characterization of packetized speech delay distributions on network links and develop criteria for packetized speech network performance. The performance criteria and delay evaluations will be used in the design of packet switched networks for voice and data - one of the alternatives in the investigation of DOD switching options.

Packetized speech traffic models and performance criteria are significant because these are the environmental conditions which provide the primary reasons why packetized speech network design differs from the data case. In general, speech traffic has a more regular and predictable arrival pattern than data and so, intuitively, one would expect the network design to be able to capitalize on this by achieving higher facility utilization with speech than is possible with data. On the other hand, network performance criteria for speech will be more stringent and, in particular, require a regularity or consistency that is not required for data transmission.

The major accomplishment reported on in this chapter is the formulation and study of a mathematical model of a single link communications channel carrying packetized voice. The extension to tandem links is formulated and is currently being implemented. The major results are:

- A unified treatment of speaker behavior models is presented. These models lead to the characterization of the distribution of an input stream of speech packets from a set of active users to a packet switch.

- A number of protocols for packet speech delivery to listener are examined and related to performance criteria.

- A computational scheme is developed for studying single link behavior under a wide variety of circumstances.

- The single link steady-state delay distribution is shown to be approximately exponential. A closed form expression

for the mean delay parameter, over a restricted range of applicability, is obtained by numerical fit.

- Standard approximations are shown to be overly conservative - they predict poorer performance at a given utilization than can actually be attained.

- One of the approximations yields a closed form expression for optimal packet length. The values predicted agreed well with results from our detailed model. Optimal packet length is shown to be proportional to the number of overhead bits per packet and otherwise depends only on the zero overhead utilization and the ratio of speaker bandwidth to link capacity. Much shorter packets than are being used in current experiments are shown to be optimal. Thus vocoders would require much smaller packets than high bandwidth (PCM) terminals. For example, the optimal packet length for eighteen 5 KBs speaker pairs on a 50 KBps link is shown to be in the 100 to 107 bit range, assuming 100 bits of additional overhead bits per packet.

- Transient behavior is investigated and it is shown that recovery to steady-state behavior from a worst-case instantaneous overutilization takes only a few seconds (3 to 5 seconds for typical parameters).

- The effects of finite buffers is investigated. A small number of buffer sustains excellent performance even at high utilization, and improves transient behavior.

- The output process is given an approximate characterization that will facilitate formulation and development of tandem link models and more general networks topologies.

- A tandem link model is formulated.

CHAPTER 5: A CIRCUIT SWITCH NODE MODEL

In this chapter we outline a generic circuit switch architecture which forms the basis of an analytic model used to obtain several performance measures as a function of relevant switch structural and operational parameters. The model possesses sufficient generality to enable an investigation of cost/performance tradeoffs encompassing different categories of switch architecture. The circuit switch model is a task in our investigation of DOD voice and data networks under circuit switching technology. The objectives of the modeling are:

- To determine circuit switch performance as a function of relevant architectural parameters.

- To quantitatively assess the impact of a particular switch architecture/operation on overall network performance.

The modeling effort is directed to obtaining the cross-switch delay and cross-network set-up delay. The model, together with switch functional specifications and technology assessment, will be used for cost modeling of the circuit switch. Solutions are obtained for the cross-network set-up delay for the following routing and signaling options:

- Network routing/control stragegies based on progressive route control and origination office control.

- Network signaling techniques based on per-circuit signaling and common channel signaling.

The model has been implemented within the circuit switched network design program described in the Seventh Semiannual Technical Report and cost/performance tradeoffs using this program are now underway.

CHAPTER 6: LARGE SCALE PACKET SWITCHED NETWORK DESIGN TRADEOFFS

In this chapter we apply NAC's design tools, developed under ARPA sponsorship, to the design of high volume packet switched data networks. This study extends previous NAC results in the following aspects:

- Significantly higher throughput range;

- Incorporation of switch and concentrator cost models into the designs.

- Investigation of various transmission to switching cost ratios.

- Incorporation of satellite channels at high traffic levels.

The base data used for the study is derived from CONUS AUTODIN II requirements which includes 272 device locations and total traffic of 1.833 MBps. The traffic range considered is 1.833 MBps to 183.3 MBps. The major conclusions of the study are:

- Line costs, particularly local access line costs are the dominant network cost component over a wide variety of hardware costs (switch and concentrator) and traffic levels.

- The optimum number of backbone sites in a terrestrial network remains virtually constant over the ranges investigated: a factor of 10 in the hardware cost, a factor of 100 in throughput level. However, the importance of selecting the optimum number diminishes with decreasing switch cost. For networks with a satellite backbone the optimum number of backbone sites is a strong function of ground station costs.

- At high throughput levels, economies of scale become less dramatic if bandwidth economies of scale are not available at multiples of 1.544 MBps. For example, the cost per 1 KBps throughput under nominal hardware cost is $139, $48, and $30, for the 1.8 MBps, 18 MBps, and 180 MBps designs,

respectively. Continuing the assumed bandwidth economies of scale to the 6 MBps range reduces the cost to $26/KBps at 180 MBps throughput.

- In comparing pure satellite vs. pure terrestrial networks at nominal costs ($1 million per ground station), the minimum cost network appears to be a pure satellite network, where the number of ground stations is 3 (vs. 5 backbone nodes for a minimum cost of pure terrestrial network).

- Reducing the ground station cost to $100,000 makes satellite access attractive at well over 10% of the nodes, and has a dramatic impact on overall network cost.

- Reducing the satellite channel cost below nominal did not alter the optimal number of ground stations but extended the range (to about 12 sites) where a pure satellite backbone is cheaper than a pure terrestrial.

- Maintaining a minimum "thin line" 50 KBps 2-connected terrestrial backbone capability in a mixed satellite/terrestrial network increases the backbone costs over the "pure" networks by about 50%.

- Backbone line costs vary approximately linearly with packet protocol overhead percentage.

CHAPTER 7: TOPOLOGICAL DESIGN OF GATEWAYS FOR PACKET SWITCHED INTER-NETWORK COMMUNICATION

This chapter presents a methodology for the topological design of interconnecting packet switched networks, procedures for realizing the methodology, and applications to two case studies. It is noted that the topological design of interconnecting packet switched networks has not been addressed before, although substantial research has been done in the areas of internet protocols, functional modules of gateways, and gateway architectures. The latter is briefly surveyed in the chapter.

One of the fundamental issues in the topological design is whether local networks to be interconnected can be upgraded, by increasing link capacities and/or topological changes. A variety of reasons including organizational, security, and ownership reasons may prevent such changes. In the chapter we address internetwork design under both options and compare the cost difference between the case in which upgrading is permitted and that in which it is not. Another issue in internetworking is whether local net traffic can traverse internet boundaries, (the merits of which are apparent from reliability and load sharing arguments). The routing algorithms used in the methodology prevent the above option.

In this chapter we:

- Formulate the topological design problem for interconnecting packet switched networks and outline procedures for its realization.

- Extend computational techniques for single network design (previously developed by NAC) to perform routing, capacity assignment and topological design of interconnecting networks.

- Formulate and implement a procedure for determining number and location of gateway halves.

The first case study involves the interconnection of four packet switched networks. Each network consists of 8 nodes, where the 32 nodes are uniformly distributed in the continental U.S.A.

Under the assumptions of constant total (intranet and internet) traffic requirements of 5 Mbps and uniform intranet and internet traffic distribution, we investigate internet costs using nominal values of $2K/m/gateway half module, and $5/mile/m 50 Kbps channel cost (one time software development cost was also considered). The following observations and conclusions emerge:

- The higher the proportion of internet traffic requirement, the higher the total system cost. (Note that the total traffic is constant). The total monthly cost for the internet ranged from $150K to $225K for 1 Mbps internet throughput, constituting 15% to 20% of total network cost. For 2 Mbps of internet throughput corresponding monthly costs were $250K to $360K, amounting to 21% to 28% of the total network cost.

- Suppose each local net is optimally designed for the local traffic, and is not allowed to change. Then in order to accommodate the internet traffic (even under the assumption that total amount of local and internet requirement is constant), gateway-halves are required at at least half of the switch locations.

- Suppose only a moderate number of gateway-halves are used. By allowing modifications on the local net link capacities (but not topologies), good internet design can be obtained.

- The total cost for an interconnected system is of the same order as the total cost for a fully integrated system, assuming that no extra protocol overhead, and no software development cost are required for the later system.

For the ARPANET-AUTODIN II case, studies were carried out assuming fixed and modifiable nets and fixed local net traffic requirements. The peculiarity of the ARPANET is that the traffic level and utilization are low but the number of switches is large. On the other hand, AUTODIN II is characterized by a small number of backbone nodes and high throughput requirements. This presents a substantially different application than the previous case study. In these studies we varied 50 Kbps channel cost from $2.5/mile/m to $10/mile/m and gateway half module cost from $2K/m to $10K/m.

The following results and conclusions emerge:

- Assuming ARPANET and AUTODIN II are not allowed to change we determined the maximum internet throughput as a function of number of gateway halves. The results shows that the maximum throughput does not increase significantly when the number of gateway halves exceeds 22. A maximum throughput of 300 Kbps can be obtained by using 6 gateway halves, 3 in each network. When the number of gateway halves is increased to 22, 17 of which are in the ARPANET, the maximum internet throughput that can be accommodated is 600 Kbps. The total

monthly internet cost ranged from \$400K to \$1000K for internet throughputs of 100 Kbps to 600 Kbps.

- Cost differences between designs with local nets fixed and designs with local nets modifiable, are sensitive to changes in unit communications cost, but less sensitive to changes in unit gateway half module cost.

- The cost difference between the two options (fixed or modifiable local nets) ranges from 1% to 12%.

- For low gateway half cost, the cost difference is more sensitive to communication cost than to internet throughput level. However, when the gateway half cost is increased, then the cost difference becomes sensitive to the internet throughput level.

CHAPTER 8: MARKOV CHAIN INITIALIZATION MODELS FOR PACKET RADIO NETWORKS

In Chapter 8 we address the problem of packet radio initialization - a major issue in Ground Packet Radio Technology under general deployment and in mobile environments. Approximation models for estimating worst case initialization time were previously developed and extensively investigated by NAC (see Seventh Semiannual Technical Report). In this chapter we develop an exact Markov Chain Model, derive its computational complexity and study the initialization time as a function of number of repeaters and transmission rates of repeaters and station. The models developed are characterized by:

- Single hop network with m repeaters under complete interference (all devices within an effective transmission distance); and a génerlized model for partial interference.

- Single common channel for all devices using a slotted ALOHA random access scheme.

- Finite and infinite station buffers for labels and random label queue management; namely, the station selects a random label from its queue each time it transmits one.

The results of the experimental investigations indicate that the:

- Initialization times are more sensitive to repeater transmission rates than station rate.

- Initialization times are not significantly increased if there are somewhat fewer buffers in the station (for initialization) than the number of repeaters.

- Initialization times increase nearly linearly as the number of repeaters increases (with the number of buffers in the station equal to approximately 1/2 the number of repeaters).

- Optimal repeater transmission rates decrease as the number of repeaters increase.

- The total transmission rates of all repeaters is nearly constant.

CHAPTER 9: MARKOV CHAIN INITIALIZATION MODELS WITH FIFO LABEL QUEUE MANAGEMENT AT THE STATION

In this chapter we model and study packet radio network initialization. The models of this chapter, as compared with those of Chapter 8, assume that the station uses a First-In-First-Out (FIFO) label queue management discipline. Apart from being a discipline more closely modeling real network operation, it is interesting to note that the complexity of the model is significantly smaller than that of models in Chapter 8. However, the model of Chapter 8 enables the investigation of minimum initialization time as a function of number of station buffers.

In this chapter we address a single hop network using a common channel and the slotted ALOHA random access scheme. For the system described we:

- Formulate a new Markov Chain initialization model based on FIFO queue management.

- Establish that the complexity of this model is $O(m^2)$, where m is the number of repeaters.

- Derive approximate solutions for this new model when the station has 1 or 2 buffers for the general interference case.

- Study numerically the solution and obtain results for the proper transmission rates of station and repeaters to minimize the initialization time, as a function of number

of repeaters, interference between repeaters and number of station buffers.

The following conclusions emerge from the studies:

- The initialization time with 2 buffers at the station is approximately 15% smaller than with 1 buffer, for the same interference pattern and network size.

- The optimal station transmission rate, q*, is nearly independent of m (number of repeaters), for both values of the buffer size; however, q* decreased 10% as we went from 1 buffer to 2 buffers. Thus, q is a function of the station architecture.

- The optimal repeater transmission rates are independent of the buffer size on the average, indicating that the repeaters need not be aware of the station's buffer structure.

- The optimal repeater transmission rate increases about 20% as interference goes from complete to zero; the initialization time decreases about 15% as we go from complete interference to zero interference.

- In both cases for large m, optimal repeater transmission rates are proportional to 1/m.

CHAPTER 1

A CLASSIFICATION OF ROUTING STRATEGIES FOR TELECOMMUNICATIONS

CHAPTER 1

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A CLASSIFICATION OF ROUTING STRATEGIES FOR TELECOMMUNICATIONS

1.1 INTRODUCTION

A Taxonomy for classification of telecommunication routing algorithms is presented. Thus far, routing algorithms were described by broad notions (or variables) such as: "fixed," "adaptive," "deterministic," "stochastic," "centralized," "distributed," etc. It appears that these broad notions are not sufficient to characterize routing in modern teleconnumication networks because of technology advances, the mixing of several classical routing algorithms, and the diversity of applications and traffic characteristics being served by one telecommunication network. For example, the degree of adaptability of an algorithm depends on the time intervals used for traffic or performance measurements and the time intervals used for updating routing tables or parameters.

The taxonomy presented in this chapter uses elementary features. Apart from enabling the classification of known routing schemes, it also suggests ways by which new routing schemes can be constructed.

The objective of this chapter is to present a methodology which enables classification and detailed characterization of routing algorithms, rather than the evaluation and comparison of algorithms. The comparison requires a criterion and a technique (e.g., analytical, simulation) for doing so - neither of which are presented. This distinguishes the chapter from survey papers that we reference, where the objective is mostly characterization of routing techniques for the purpose of comparison under some network and traffic characteristics.

1.1.1 What is Routing for Telecommunications?

For our purposes, a routing strategy will be a method for selecting a physical path along which information is to flow. Routing arises in many different applications. In each of these applications, the actual physical constraints have a very strong

bearing on the types of solutions one can propose and implement. Because of these differences, routing strategies for one situation are often very inappropriate for others. For example, routing problems occur in the design of wiring and printed circuit boards for electronics. Here, typical constraints are: the avoidance of contiguity of channels or wires, the restriction of paths to a plane, and the avoidance of edge effects. Few of these constraints have counterparts in routing problems that occur for vehicular traffic or telecommunications. On the other hand, the problems that arise in routing of vehicles and in routing of signals and messages in communication systems, are much closer, and much of the literature in either field applies to the other. Indeed, it is apparent that much research has gone on in parallel without the awareness of duplication of effort.

We will focus our attention on routing problems and strategies for telecommunication systems; that is, systems whose primary function is the transmission of digital or analog information in the form of electrical signals through a system of connecting links and intermediate points.

1.1.2 Examples of Routing in Telecommunication Systems

We will be concerned with routing in any and all telecommunication systems. To give an idea of the scope of this undertaking, we will give three examples of actual routing schemes presently in existance; the routing in the ARPANET, the AT&T Network and the AUTOVON System. We first establish a brief outline of switching techniques to place these specific schemes in context. The classical techniques are circuit and message switching. A newer established method is packet switching. A proposed method still under investigation is hybrid switching.

Circuit Switching

The key aspects of the circuit switching scheme are:

1. A message is not sent until an end-to-end path is established.

2. Messages are not stored at intermediate nodes.

3. The routing problem consists of the selection of the end-to-end path or, in other words, the determination of the route for the system signals.

Message and Packet Switching

The key points in packet switching are: (a) messages are broken into "packets" of a fixed maximum length; (b) packets are sent before an end-to-end path is established; (c) packets are stored at intermediate nodes until links become available. If criterion (a) is dropped, then the system becomes what is classically known as a message switched system. An example of a message switched system is the Western Union Telegram System. ARPANET is an example of packet switching.

Integrated Switching

An integrated system retains some of the properties of more than one of the "classical" switching techniques, circuit, message and packet. For example, some messages may be sent in a packet switched mode whereas others may be sent in a circuit switched mode. True integrated systems do not exist at present. A proposed system [FISCHER, 1976] allocates a variable number of time slots in a frame to circuit switching, and the remainder to packet switching. Determination of

messages suitable for packet or circuit switching may be a function of many variables such as message length, delay requirement, and network congestion.

The ARPANET is a packet switched system. When messages arrive at an input point, namely Interface Message Processor (IMP), the message is broken into packets of a predetermined maximum size. These packets receive headers which contain destination addresses. The packets are then sent to intermediate points where they are stored until a link becomes free for further transmission. The packets in a given message can make their way independently through the network until they reach the destination IMP. If they arrive out of order, they must be reassembled at the IMP. In the ARPANET, the actual decision as to which link to pick next is made locally at each IMP based upon update information distributed from its neighboring IMP's. Each IMP has a table which contains an estimate of the time it would take to deliver the packet to each destination, depending upon, on which link the packets were sent. Periodically, each IMP receives estimates from its neighboring IMP's on delay times from those IMP's to all destinations. Combining these estimates with its own estimates, each IMP then updates its routing tables. If the tables are updated fast enough then this algorithm is a distributed version of Bellman's Shortest Path Algorithm for finding shortest paths between all nodes in a network.

The AT&T Switching System [WEBER, 1962], is an example of a circuit switched system. In this system, as in Figure 1, there is a hierarchy of nodes in which a node higher in the hierarchy is not used unless the lower level node is blocked. This has the effect of maximizing the probability of finding a route for the next call arrival - since higher level nodes can serve a larger user population. To further simplify routing, links are ordered so that certain links became "high usage links," and others, alternates. As an example in Figure 2, for a call from Syracuse to Miami, the customer dialed a 10 digit code at Syracuse. The various routes which might then be tried are shown. The links are numbered in the order of trials.

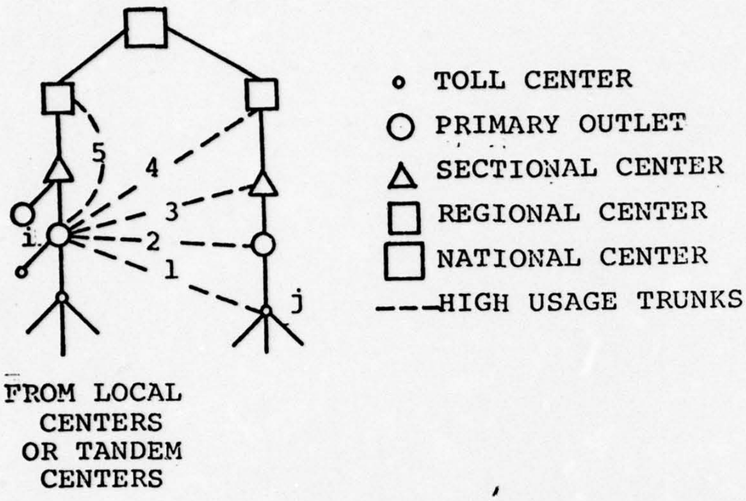

FIGURE 1: ROUTING IN THE AT&T HIERARCHICAL NETWORK: THE NUMBERS ON HIGH USAGE TRUNKS INDICATE THE FIXED ORDER OF ALTERNATE LINKS FROM i TO j

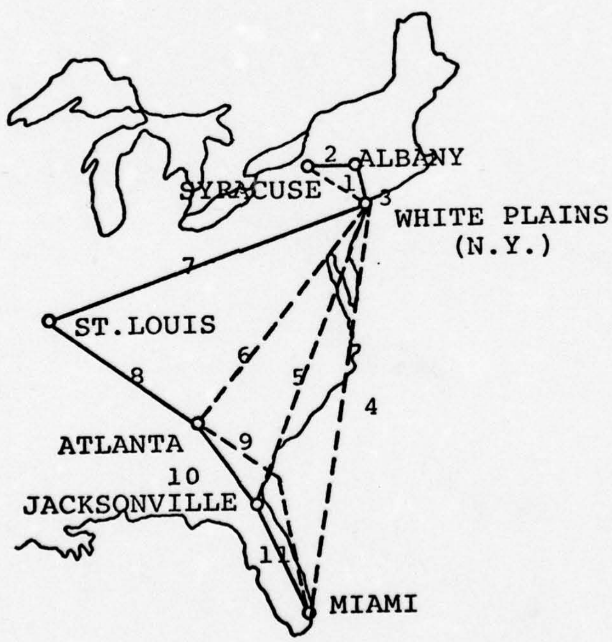

FIGURE 2: AT&T NETWORK: EXAMPLE OF ORDER OF LINKS IN ESTABLISHING A PATH FROM SYRACUSE TO MIAMI

In this example, a maximum of 11 circuits could be tested for an idle path. Dotted lines show high usage routes which, if found busy, overflow to the final routes represented by solid lines. The switching equipment at each point, upon finding an idle circuit, passes on the required digits to the next switch.

The distinguishing characteristics of the AT&T routing method for circuit switching are:

1. There is a fixed order of alternate links.

2. Not all possible paths are allowed.

3. The routing establishes a path which utilizes the lowest level of switching centers in the available hierarchy.

Many variations can be designed based upon the circuit switching and packet switching schemes. An example of another approach to circuit switching is the AUTOVON Network. In contrast to AT&T's direct dialing hierarchical network where routing has to follow rigidly determined paths through the hierarchy, AUTOVON's Polygrid Network provides for alternate routing around disabled centers over many possible paths. The AUTOVON Network has been designed with a particular symmetric structure which will allow for great reliability and effective symmetry in routing decisions, as well as the availability of many alternate routes. All nodes are of equal rank ("nonhierarchical") and routing decisions are controlled by automatic switching equipment at each node ("progressive routing"). The structure consists of a repeated set of hexagonal structures, as shown in Figure 3, with their superposition, shown in Figure 4, and with long haul trunks added as in Figure 5. Each switching center has the capability to search on each call, one direct trunk route to the called destination, if available, and two additional

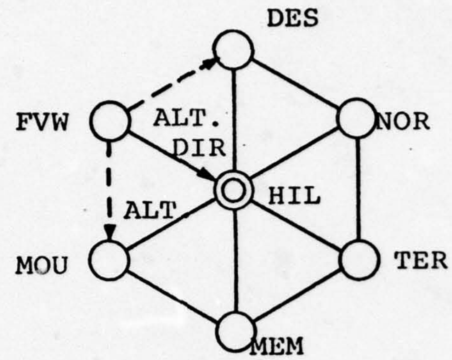

FIGURE 3: HEXAGONAL STRUCTURE AND NONHIERARCHICAL ROUTING IN THE AUTOVON NETWORK: FIGURE SHOWS DIRECT AND TWO ALTERNATE ROUTES (NOT ORDERED) FROM FVW TO HIL

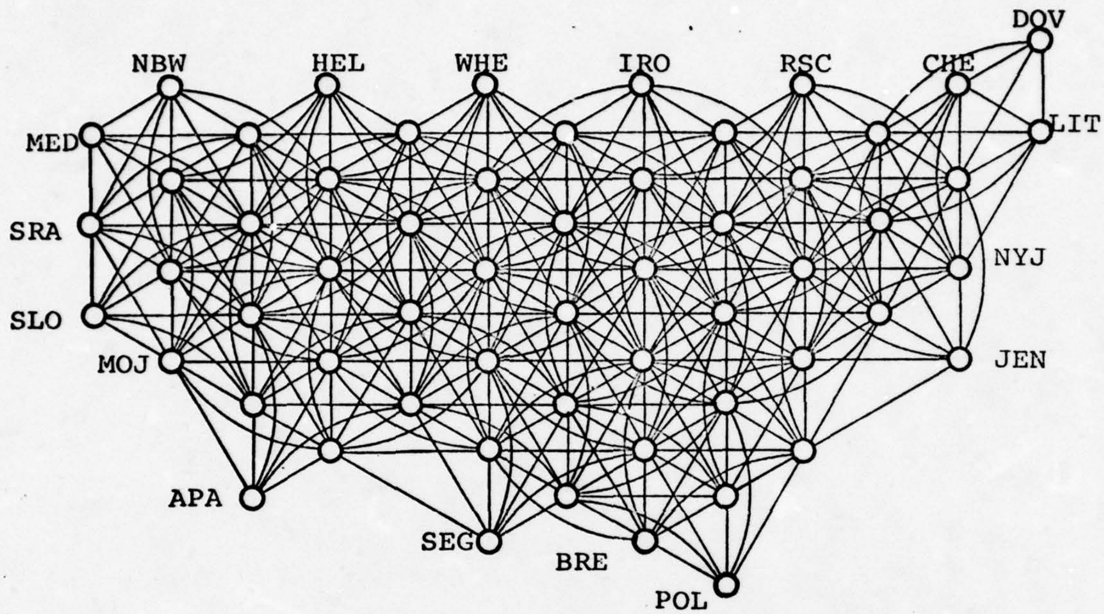

FIGURE 4: NONHIERARCHICAL STRUCTURE OF THE AUTOVON NETWORK

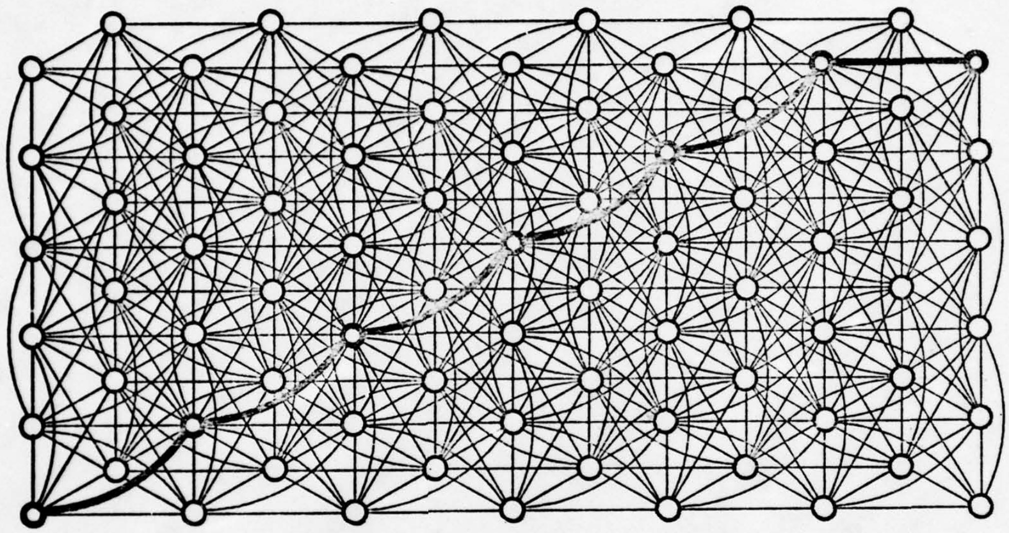

FIGURE 5: LONG HAUL TRUNKS ADDED TO AUTOVON NETWORK STRUCTURE

links to other switching centers. The links are selected for ultimate route advance. Thus, a link cannot be picked to a node already in the route. Furthermore, there is a maximum number of links in a route.

The dominant characteristics of the AUTOVON circuit switching method are:

1. The network is nonhierarchical.

2. Many alternate routes available.

3. No ordering of links.

1.1.3 Types of Routing Problems

As we have already seen, in some cases routing problems concern the routing of "messages" through the network between sender and receiver. In other cases, they concern the routing of system "signals" in order to establish paths. Other distinctions must also be made. Some routing problems arise in the analysis and design of telecommunication systems. Other problems arise in the implementation of routing schemes. The solutions to the problems depend upon the type of network we are considering - message switching, packet switching, circuit switching, or hybrid - and the objectives of the routing scheme.

A number of distinctions which are ordinarily made in categorizing routing problems will not enter our discussion. For example, the distinction between voice and data networks is, indeed, important in determining requirements on throughput and delay. However, either or both can be handled by routing methods falling into our categorization. Similarly, we ignore the distinction between point-to-point and broadcast modes, and distinctions between mathematical models such as single terminal flow problems and multiterminal flow problems.

1.1.4 Objectives of Routing Methods

There are many ways to set goals for routing methods. For example, the network owner may have the objective of maximizing his profit while satisfying user minimal requirements. Another objective might be to provide the "best service" to the user in an existing network. In vehicular traffic systems, there is an obvious difference between the system "controller's" objective of optimizing average delay and individual user's goal of minimizing his delay. In telecommunication systems, the user usually cannot control his own route [HARRIS, 1973]. Actually, Kleitman [CLAUS, 1974] makes the argument that tariffs should be set so that these two objectives become identical. Nevertheless, providing "best service" is itself a complicated objective depending upon priorities and differences in requirements. For example, in some implementations of a network carrying both voice and data, improvement of performance for one type of traffic might degrade performance for the other. With these caveats in mind, we will adopt a single generalized objective throughout the development. The objective is the most efficient design and utilization of the network to guarantee the user at least a minimum level of performance. The complexity and interest in the problem, of course, arises from the vast variety of valid interpretations which can be presented for "efficient system design and utilization" and "user level of performance". Among the measures are:

1. Throughput, delay, blocking probability, reliability.

2. Adaptability to variations in traffic level and statistics.

3. Adaptability to outages.

4. Cost of implementation.

Obviously, in view of these very complex objectives and the many tradeoffs between them there is no single best routing scheme. Indeed, for a given application, at different times different routing schemes may be preferable. Even where detailed simulations of routing schemes have been done, the results of the comparisons must be interpreted very carefully in order to avoid arriving at decisions which are incorrect. Quite often the results of comparisons of routing strategies depend upon particular simplifications in the models or upon very particular constraints on the networks, which prevent their general applicability. On the other hand, with most routing schemes, quick intuitive appraisals can often be made of their general properties in terms of the above objectives. For example, only a qualitative understanding is necessary to explain the merit of ARPANET packet switching for terminal traffic - efficient utilization of links and reliability, because of distributed decisions. Similarly, it is clear that for long file transfers or long voice conversations, circuit switching is a strong candidate since long setup times can be tolerated and dedicated facilities for long periods are not wasteful. It can be easily seen that, because of many more potential alternate routes, an AUTOVON type circuit switching method could be designed with more options for route selection in case of link or switch failure than for AT&T type circuit switching method. Thus, even though the precise break-even points must be determined by detailed analytic and simulation models, a great deal can be determined by a general understanding of existing and potential routing schemes.

1.1.5 Purpose of Classification

This brings us to the motive classifying routing strategies for telecommunications. In order to be able to evaluate existing routing schemes and to devise new ones which have the properties we desire for given objectives, we must be able to categorize the rout-

ing schemes so that their properties become evident. Accordingly, we have the following five goals in presenting a classification of routing strategies:

1. To facilitate the understanding of routing strategies and the relationships among them.

2. To present routing in a framework which is not specialized to circuit, packet, message or hybrid switching.

3. To present a framework which is general enough to include all existing routing schemes.

4. To enable us to formulate new routing methods.

5. To present a framework which will allow the simulation of a variety of routing schemes.

In order to accomplish these goals we will try to present a classification according to the following methodology. We will formulate a basic set of broad questions about routing. These broad questions will form the main outline of the classification. The answers to the questions will form the remaining part of the classification and will satisfy the following properties: (a) The answers will automatically classify all existing methods; (b) The answers will automatically give a variety of possible new routing methods; (c) The questions and answers will be such that they could correspond to decision points in analytic or simulation programs to model the routing schemes.

1.1.6 Previous Classifications

A number of classifications of routing schemes have been given before. These classifications are useful in the context of their original motivations and findings.

1. [PROSSER, 1962a, 1962b]. Prosser was interested in the distinction between military and civilian communication systems and presented the concepts of random and directory routing.

2. [BOEHM, 1969]. Boehm and Mobley followed up on the original pioneering work of Baran [BARAN, 1964] on distributed communication systems and introduced the concepts of adaptive stochastic and adaptive deterministic routing.

3. [FULTZ, 1971]. Kleinrock and Fultz gave a classification of routing schemes which would be useful for the simulation of packet switching systems.

4. [McQUILLAN, 1974]. In his doctoral dissertation McQuillan gave a classification which would be useful for the implementation of routing schemes in the ARPA network.

5. [GERLA, 1973]. The classification in Gerla's doctoral thesis was aimed at improving the implementation of schemes for the ARPANET.

6. [RUDIN, 1976]. Rudin was also concerned with improvement of implementation schemes for packet switching networks and his delta routing evolved from his classification of these schemes.

7. [BAHR, 1974]. Bahr and Majithia were concerned with classifications for the simulation of packet switching strategies.

These previous schemes are included, in terms of the terminology they introduced and used, in Appendix B in outline form. It is not our purpose to relate or discuss in any detail these previous classifications. We have the advantage of hindsight over previous classifications, we have more models and examples to include, and we have a more general purpose in mind. One of the merits of our classification will be that the terminology introduced in these previous classifications will become clear as they appear in our classification.

We also will differ from some of the above classifications in two particular respects. We will not include flow control schemes or shortest path algorithms in the classification. Quite often flow control schemes and shortest path algorithms are very difficult to separate from routing methods and some of their terminology will, of course, impinge upon the classification. For example Lam, [LAM, 1976] considers buffer allocations needed for various routing methods. But, wherever possible, they will be separated from the routing schemes. Flow control methods usually deal with the protocols for handling individual messages and packets in order to control the number of messages allowed in the system and to handle them properly for bookkeeping and error detection. Thus, flow control deals with the handling of acknowledgements, storage allocation, message entry permission, the assignment of priorities, and so on. Of course, if a message is discarded after it has been in the network for too long, then this is a flow control issue. It can also be viewed as part of the routing scheme since essentially part of the route is eliminated for a message. Similarly, shortest path algorithms are used quite often in the implementation of particular routing schemes. The complexity, speed and robustness of these algorithms often determine the feasibility or merits of a particular routing scheme. Nevertheless, the shortest-path algorithm used is a detail of the routing strategy rather than a determinant of the strategy itself.

1.1.7 Outline

In the next section we present our classification of routing methods. Many papers have appeared proposing different routing schemes. These schemes should appear naturally as part of the classification. Existing methods will be referenced in the classification by the names given by their inventors and by their designation in the bibliography. Our purpose here is not to propose optimal methods. Hence, we will not discuss analytic and simulation results achieved by the various authors. The referenced papers contain these arguments.

1.2 CLASSIFICATION OF ROUTING SCHEMES

Almost every classification of routing schemes that has been done, at least implicitly, proceeds on the basis of formulating general routing questions. Rudin [RUDIN, 1976] bases his routing classification on three major questions relating to routing strategy, routing decisions and routing information. Our classification consists of three main headings which result in nine fundamental questions, as follows:

1. Speed of path selection, signals and messages:

    1.1 How fast is the routing method in establishing a new path relative to message delay requirements?

    1.2 What are the signal delays?

    1.3 What are the message delays relative to the signal delays?

2. Measurements:

    2.1 How often are measurements made and routes updated?

    2.2 What is measured?

    2.3 Where are measurements sent to (i.e., which are "informed nodes")?

3. Decisions:

    3.1 Where are decisions made?

    3.2 How is a route established?

    3.3 How is a route terminated?

The answers to these questions form the outline of a classification of routing methods. This outline is given in Appendix A. The classification can stand alone in that the answers to questions define routing schemes regardless of what names we give them. In Appendix A we have included some of the standard names associated with various algorithms such as, "adaptive," "random," "hot potato," and so on. In some cases these names are suggestive and clearly identifiable in terms of their location in the outline. In other cases, such as for fixed or deterministic or stochastic routing, the terminology used by various authors is inconsistent. The location of these schemes in our outline should clearly define their scope as we intend them. In this section we will give the classification exactly as it is in Appendix A, with the same numbering, but we will, in addition, include in the body of the outline various comments indicating the history of some concepts, their interrelationships and their origination in terms of various references. We will not indicate in the outline every single place where a given reference is relevant. We will merely try to flesh out the outline with an intuitive background by indicating various representative types of algorithms. The classification in this section and in the Appendix is given in terms of the following notation.

NOTATION

NI, NJ	Nodes
NS	Source Node
ND	Destination Node
(NI, NJ)	Link between NI and NJ carrying information from NI to NJ
L(NI, NJ)	"Length" of (NI, NJ) in miles, dollars, etc.

NOTATION (Cont'd)

L(NS, ND)	Length of a path between NS and ND.
TI, TJ	Time
TM	Time between measurements.
TU	Time between routing updates.
MI, MJ	Measurements at NI, NJ.

1.3 SPEED OF PATH SELECTION, SIGNALS AND MESSAGES

1.3.1 How Fast is The Routing Method in Establishing a New Path Relative to Message Delay Requirements?

The primary purpose of this question is to classify routing methods which are applicable for implementation or design. There are several important points in the phrasing of the question.

First the question does not itself refer explicitly to implementation vs. design. The reason is that a given routing method may be used either for implementation or design depending upon many factors. For example, the routing method implemented in the ARPANET can also be used to study routing in order to design new networks.

Second, by focusing on the speed of the routing method in establishing a new path, relative to delay requirements, we have the key parameter which determines whether a method can be used for implementation in a given environment. If the node processing time of an algorithm is small compared to the delay required for the message (delay may be setup time, transmission time, queueing delay, propagation delay or combinations of these) then it may be used for implementation. If not, then it can be used only for design. Frank and Chou [FRANK, 1972] propose linear programming formulations to yield flow distributions in a packet switched network. On most machines, the running time is too long for this method to be considered for implementation when packet delays of a few tenths of a second are required. On the other hand, the method is very useful for establishing bounds on design parameters. In the same paper, various heuristic methods using shortest path methods are also introduced for design. However, variations of these methods with proper flow control procedures could also be implemented.

Third, we speak about a routing method "not a routing algorithm." An algorithm is a mathematical procedure, whereas a routing method implies an algorithm implemented in hardware and software. Clearly, there is a tradeoff between speed and system cost for

hardware requirements in memory, cycle time, etc., and software complexity. Hence, it is the actual implementation which determines whether a routing method can select a path in time for a message to be transmitted within its delay requirements.

1.3.1.1 Slow

Slow algorithms are those that if implemented in a network, the sum of processing times of nodes along a path is larger than the delay requirement for the message. Slow algorithms can be very useful if they are reasonably accurate and model a number of phenomena which occur in real systems. They can bound actual designs and can be used to obtain good first pass designs for networks. Examples of these are: [LIN, 1975] for AUTOVON, [KNEPLEY, 1973] for circuit switching, and [KATZ, 1967] for circuit switching. Often the failure to make this distinction between slow and fast algorithms results in confusion by authors in attributing merit to various algorithms [CHYUNG, 1975].

1.3.1.2 Fast

Among the fast algorithms are those which can be implemented in physical networks to route signals and messages, and those which are fast enough to be used in interactive network design procedures. That is, they can be run hundreds or even thousands of times while varying the topology of a network in design iterations. The design algorithms, of course, must be shown to produce results and designs which model performance tradeoffs well enough to have confidence in the resulting designs. Furthermore, an algorithm for design should produce networks which perform well when used with routing algorithms which are implemented.

Some routing methods may be "fast" or "slow" depending upon the properties of the network for which they are to be used. For example, a routing algorithm used in some routing schemes [GOTTO, 1973] does a time consuming matrix inversion once, but then responds very quickly to selecting new paths, so long as switch or link outages do not change network topology. Hence, a method based on the algorithm may be "fast" for a network with few outages but slow otherwise.

1.3.1.2.1 Iterative Design

Among the most useful iterative design routing methods are, [CANTOR, 1974], [CHOU, 1972], [FRANK, 1971], [FRANK, 1972], [FRANK, 1970] and [FRATTA, 1973].

1.3.1.2.2 Implementation

Actual route implementation algorithms include: [GORGAS, 1968] for AUTOVON, [McQUILLAN, 1974] for ARPANET, and [GITMAN, 1976] for the Packet Radio Network.

1.3.2 What are the Signal Delays?

We make the distinction between a "message" which contains information to be sent from sender to receiver, and a "signal" which is information generated and sent by the network in order to establish route.

1.3.2.1 Zero Seconds

For routing methods in this category, the information carried by signals for routing are received instantaneously at their destinations.

1.3.2.1.1 Ideal Observer

If signals take zero seconds to make their way to "control points" then one explanation is that there is an ideal observer who has a complete instantaneous view of the network. This occurs only in design situations. However, routing strategies, based upon an ideal observer, can approximate actual flow in real networks. Examples of these are, [CHOU, 1972], [FRANK, 1971], [FRATTA, 1973]. The above methods use flow techniques along shortest paths in order to achieve maximum flow. Other authors [SCHWARTZ, 1975], [SCHWARTZ, 1976], [SEGALL, 1975] use differential equation approaches for state space models which also assume instantaneous information.

1.3.2.1.2 Isolated

The other possibility for zero signal delay is that there are no signals. These are schemes which can be implemented and in which decisions are made at individual nodes on an isolated basis, either at random or based on measurements at each node.

1.3.2.2 Link Delays

The model here is that the maximum delay in sending any signal is the delay incurred on the single links. This can include many implementations of circuit and message switching. Since there is no queueing delay incurred at intermediate nodes, the model is that inband signalling is used - hence there is no contention between signalling messages.

1.3.2.3 Link Plus Queueing Delays

This is the same as 3.2.2 except that common channel signalling is used; hence there is contention between signalling messages which results in queueing and transmission delays.

1.3.3 What are the Message Delays Relative to the Signal Delays?

The question here is, how far behind the signal is the message? Does the signal travel a single link before the message follows it on that link? Does the signal travel two links before the message follows it? Does the signal travel the entire length of a path between source and receiver and establish the entire path before the message follows it? Finally, is there zero delay between the signal and message? That is, is the message its own signal? The answers to these questions form a very simple and natural categorization of circuit, packet, message, integrated, and other switching techniques which comprise the most fundamental differences among routing methods.

1.3.3.1 Zero Delay

This corresponds to some implementations of packet and message switching. In these schemes, information is sent without any signals. However, not all implementations fit this category. For example, in ARPANET, allocation requests for buffer space at the destination switch must be made for multipacket messages. In essence a "virtual path" is established by the signal which introduces a delay. In most analysis and design algorithms, zero signalling delay is assumed. The pioneering work describing packet switching with zero signalling delay and the "independence assumption" made for analysis is [KLEINROCK, 1964].

1.3.3.2 One Link

This corresponds to reservation packet or message switching. A signal is sent over a link and perhaps an acknowledgment is returned before a packet is sent. A scheme of this sort has been proposed for packet data transmission over cable television systems [FRISCH, 1975].

1.3.3.3 L(NS, ND)

The entire path is established before a message is sent and is used in circuit switching [GORGAS, 1968], [WILKINSON, 1956].

1.3.3.4 Variable Length

The choice is made by the network or user depending upon various factors such as the source node, the terminal node, the time, the traffic level and various other parameters. This is integrated switching. Various studies are under way now to determine methods of implementing integrated networks. One scheme involves time division multiplexing such that in each time frame there is a movable boundary which determines how many time slots are used for packet switching and how many are used for circuit switching. In a cruder version one can simply consider manual or automatic switching from a packet switched mode to a circuit switched mode. Similarly, one can also consider geographically inhomogeneous networks. For example, it is possible to have different networks each of which has a different type of switching. When these networks are connected by "gateways" the result can be designated an integrated network.

1.4 MEASUREMENTS

The timing, nature and use of measurements in routing is one of the most difficult and important issues to be faced. The measurement program is closely tied to the path selection algorithm. If the algorithm uses global information, more measurements are needed; similarly, a higher frequency of measurements is needed when an algorithm requires faster adaption to variations in traffic load and patterns. It is important to note that propagation of measurements may utilize a significant amount of network resources for information transmission. The result might be an infeasible or unstable routing algorithm. The answers to measurement questions can make the difference between the success or failure of a routing scheme and there is a wide variety of approaches.

1.4.1 How Often are Measurements Made and Routes Updated?

The distinction is made between the timing of measurements and routing updates. Often these are identical; whenever a measurement is made, routes are changed. The route changes can be made via routing tables or biases in decision criteria or various other ways. The answers to the measurement questions define very precisely the difference between a number of existing and proposed routing schemes that have appeared often in the literature.

1.4.1.1 TM = TU = ∞

This means that there are no measurements for routing updates. This corresponds to deterministic routing [BOEHM, 1969].

1.4.1.2 TM, TU Finite

This means that routing measurements are made and that routing is updated. It corresponds to stochastic routing [KLEINROCK, 1964], [BOEHM, 1969].

1.4.1.2.1 TM vs TU

The general question is the relative values of measurement and update times.

1.4.1.2.1.1 TM < TU

In some cases, a decision may be made not to update routing tables unless measurements indicate some minimum threshold of variation in traffic or topology.

1.4.1.2.1.2 TM = TU

In this case, routing tables are updated whenever measurements are made. This is the primary scheme used in the ARPANET [McQUILLAN, 1974]. It is also used in most circuit switching approaches where the measurement that is taken is whether a trunk or circuit is blocked. In ARPANET the measurement is the delay on links. In a packet switched network if TM and TU are short enough, then packets from the same message could be sent on different routes. This is a form of bifurcated routing.

1.4.1.3 Uniformity in Space

1.4.1.3.1 TM (NI) = TM (NJ) for all NI, NJ

In this case, the same measurements are made everywhere in the network.

1.4.1.3.2 TM (NI) $\neq$ TM (NJ) For Some NI, NJ

It is possible to change the frequency of measurements for different areas of a network depending upon the past history of local congestion. Similarly, if different networks are connected by "gateways," then the frequency of measurements can be different in the different networks. In packet radio, routing measurements can be sent at different rates by different nodes depending upon the location and mobility of the node.

1.4.1.4 Uniformity in Time for Measurements

1.4.1.4.1 TM(TI) = TM(TJ) for all TI, TJ

This is called synchronous measurement.

1.4.1.4.2 TM(TI) $\neq$ TM(TJ) for some TI, TJ

In this case measurements may be event driven and depend upon the history of measurements or other external information.

1.4.1.5 Uniformity in Time for Updates

1.4.1.5.1 TU(TI) = TU(TJ) for all TI, TJ

This corresponds to synchronous updates and is the mode of operation in the ARPANET, even though the update times are not the same instant, i.e., they are asynchronous.

1.4.1.5.2 TU(TI) $\neq$ TU(TJ) for some TI, TJ

BBN [BBN, 1975] has recommended that the frequency of update depend upon whether the estimated delays have increased or decreased.

They call this "hold down." Boehm and Mobley [BOEHM, 1969] use the same idea and call it "negative reinforcement." Their goal is to prevent looping. BBN's goal is to prevent excessive changes in routing which do not improve routing performance. Both of these fall under the category of asynchronous updating.

Of course, there are many different possible answers to the above questions on measurements and in many cases they can be used to define completely new routing schemes. For example, the updating can be done on a regional basis which is predetermined on the basis of projected traffic. The boundaries between the regions can be updated on a daily or monthly basis or any other period which is appropriate. The result is a new routing scheme which fits naturally into the classification and which then has properties that can be readily determined on a qualitative basis from the location in the classification.

1.4.2 What is Measured?

1.4.2.1 Outage

Prosser [PROSSER, 1962a] suggested possibly just measuring network topology. Similarly, in [NAC, 1973] it was suggested to simply store a picture of a network at a node.

1.4.2.2 Traffic

This is the most common measurement in practical networks. In circuit switching networks, trunk occupancy is measured. Given this decision it is still open to question as to the frequency of sampling, or how samples are stored, and how they are processed.

1.4.2.3 Delay

Delay is measured in the ARPANET and path delay is estimated. Boehm and Mobley [BOEHM, 1969] suggested measuring delay on inward

links only and using these to estimate the delay on outward links. The measurements are simple but can be misleading. Their name for this scheme is called "backward learning." When they combine backward learning with "negative reinforcement" they call it "biadaptive" routing. [NAYLOR, 1975] makes use of similar measurements in order to avoid double link looping or "ping-ponging."

1.4.2.4 Packet or Message Parameters

It is possible to put tags on packets or messages which change as the message progresses through the network. Information in these tags is a measure of the history of the message. Various network parameters can be estimated, based on the tags.

1.4.2.4.1 Number of Hops

The utilization of paths with an upper bound on the minimum number of hops can easily be implemented using these tags [CHOU, 1972]. For example, messages having traversed too many hops may simply be discarded.

1.4.2.4.2 Delay

A more complicated mechanism is required to keep track of an actual packet delay. For the number of hops, all that is required is a counter which is updated by each node. To keep the delay on each packet, of course, the delay must be measured on each hop.

1.4.2.4.3 Route

It is, of course, possible for the message to keep track of the path that it has followed. This can be partial information or complete information.

1.4.2.4.3.1 Partial

[NAYLOR, 1975] has proposed keeping track of one node to avoid ping-ponging. For example, in packet radio the packet ID is "memorized" and is used to prevent accepting the same packet as was sent [GITMAN, 1976]. The time that the ID is kept "memorized" implies the size of the loops to be prevented.

1.4.2.4.3.2 Complete

The only published account of an attempt to keep track of a complete path in a header is in the packet radio application [GITMAN, 1976].

1.4.2.4.4 Nothing

If no measurements are made we have either fixed [KLEINROCK, 1964] or random [PROSSER, 1962a] routing.

1.4.3 Where are Measurements Sent to (i.e., Which are "Informed Nodes"?)

Once measurements are made, they can be used locally or they can be sent to different points in the network. This is one of the most crucial aspects of routing since the answer to the question determines, to a large extent, the capabilities required in the network nodes, the reliability of the network, its speed of adaptability to change, and its throughput and delay. For example, if all information is sent to a central node to be acted upon, then the reliability may be dominated by the reliability of the central node. Furthermore, it takes time to send information to the central node and hence the network may be unresponsive to rapid changes. On the other hand, once the central node has the information, it can make the most intelligent choice of routing and use the network effi-

ciently. At the other extreme, decisions can be made locally with very little information, in which case routing is done almost randomly. This makes very inefficient use of the network in that a message will traverse more links than are necessary. But, on the other hand, it is extremely reliable in that decisions can be made even when many switches and links are down. Furthermore, signalling overhead is zero. There are all sorts of intermediate solutions which have been proposed and implemented. Furthermore, by posing new answers to these questions a whole new hierarchy of network routing schemes can be developed.

1.4.3.1 All Nodes

In this scheme the routing information is sent to all nodes which may make decisions on a distributed basis. One such scheme for packet switched networks has been proposed in [GALLAGER, 1976a, 1976b].

1.4.3.2 A Proper Subset of Nodes

This amounts to either sending to a single node which corresponds to centralized routing data base, or sending information to more than one node which corresponds to distributed routing first introduced by [BARAN, 1964].

1.4.3.2.1 kth Order Neighbor

[PROSSER, 1962b] uses kth order neighbor knowledge to assume that a message will automatically home in on a destination node on a predetermined path once it arrives within k-hops of that destination.

1.4.3.2.1.1 Nearest Neighbor

One of the most common methods of sending measurements in actual implementations is the nearest neighbor approach. In the ARPANET delay estimates are sent from every node to its nearest neighbor. Various modeling schemes have also been devised that require only nearest neighbor information. [STERN, 1976] has devised a relaxation method for solving a state space optimization formulation of the problem requiring only information from nearest neighbors. [BUTRIMENKO, 1964] also does a shortest path approach on a bidirectional basis using nearest neighbor information.

1.4.3.2.2 Control Nodes

Certain nodes called "control nodes" can be selected to receive routing information.

1.4.3.2.2.1 One Central Node

This is called the Network Control Center (NCC) approach or Network Routing Central (NRC) approach.

1.4.3.2.2.2 Source, Destination

[NAC, 1976] has suggested an approach in between the distributed and the central routing approach. In this scheme the source for a given message receives the information on delay and it establishes the path for messages originating from it. This is a compromise in terms of amount of information that must be transferred and in terms of vulnerability to network control center damage. In a circuit switched network with originating station control, all measurements are sent to the originating station.

1.4.3.2.2.3 Other Set of Control Nodes

It is possible to envision that gateway nodes or regional centers receive routing information [GERLA, 1973]. As a matter of fact, depending upon the geography and congestion (such as on "Mother's Day"), many different subsets of nodes can be delegated routing responsibilities.

1.4.3.2.3 Nowhere

This corresponds to isolated routing in which no routing information is sent. In this case local nodes can only act on the information they measure. One method [KLEINROCK, 1969] "shortest queue plus bias" allows a link to be selected on the basis of the shortest queue of messages for a link plus a bias in terms of estimated delay. If the bias is set to zero then the scheme is called "hot potato" routing [BARAN, 1964], [BOEHM, 1964].

1.5 DECISIONS

Routing decisions are often made at the nodes to which information is sent. However, this is not necessary and the decisions can be relegated a priori, or based upon measured information, to a subset of nodes. Furthermore, the types of decisions and their impact are strong determinants of routing methods.

1.5.1 Where Are Routing Decisions Made?

1.5.1.1 At Informed Nodes

This is a common scheme in that those nodes which receive information are usually the ones which perform the routing decision, ARPANET is an example.

1.5.1.2 At a Proper Subset of Informed Nodes

This is another common situation in which nodes may be used as relays for information, but only specific nodes such as the source or destination or the network control center performs the actual routing decisions. Similarly, in a hierarchical structure the nodes may act as intermediate points, but only regional switching centers may act as decision points. Another example of this type of structure is one of the algorithms for directed broadcast packet radio [GITMAN, 1976]. In an originating station controlled circuit switched network, the signalling information is sent to the originating station where the path is established.

1.5.1.3 Either All Informed Nodes or A Subset

An example of this is the δ routing of [RUDIN, 1976]. Information is sent to a central routing site. However, if the change in

traffic is less than some parameter in magitude, the central site will defer to the local site for a decision; otherwise, it will make a decision. Many other variations are possible depending upon structure of the network. For example, decisions might be made at gateways or at internal points in the network depending upon various parameters. Also, in hierarchical algorithms for packet radio, local alternate routing is done by local nodes, whereas complete path changes are done by a central node.

1.5.2 How is a Route Established?

By this question we do not mean what shortest path algorithm is used, but rather the nature of the path which is being sought and the manner in which it is sought.

1.5.2.1 Number of Paths Selected Simultaneously

Of course, the number of paths used simultaneously determines very much the classes of algorithms which must be considered. However, it also relates critically to various other parameters in the network such as, reliability and throughput.

1.5.2.1.1 More Than One Path

In the most complex situation many paths may be picked at a time. If this is the case, then there are still a number of options in the sense that paths may be picked simultaneously for simultaneous transmission of the same message or for different messages, or the paths may be used in some fixed order depending upon various contingencies.

1.5.2.1.1.1 Type of Redundancy

1.5.2.1.1.1.1 Duplicate Messages Sent Over Different Paths

This is called "flooding" or "broadcast" mode and is, for example, a natural mode for packet radio. Another possibility is sending linear combinations of messages in order to achieve redundancy. This has been used by [MAXEMCHUK, 1972] and is called "dispersity routing."

1.5.2.1.1.1.2 Different Messages Sent Over Different Paths

In this mode of operation, many different paths may be used simultaneously for different messages or different packets in the same message. Usually these schemes are based upon maximum flow approaches to multicommodity flow problems. The cut saturation algorithm [CHOU, 1972] is an application.

1.5.2.1.1.2 Two Paths

This is such a common special case that we can try to characterize the nature of the paths which are selected.

1.5.2.1.1.2.1 Diverse Paths

It is often the case that the paths which are specified are required to have no branches in common or no nodes in common. An example of this is "Bifurcated Routing" [HSIEH, 1976] sometimes called "Split Routing." The situation also arises in mixed terrestrial satellite networks where two diverse paths may be established, one via satellite, and one via ground links [HUYNH, 1975], [HUYNH, 1976].

1.5.2.1.1.2.2 Partially Overlapping Paths

It is sometimes easier to specify algorithms which yield paths with some level of diversity but not complete diverstiy. These paths still maintain some of the reliability properties of diverse paths. [CEGRELL, 1975] has used a method of switching among sub-networks to pick links for the path in the TIDAS Network.

1.5.2.1.2 Single Path

A common situation in many networks, such as the AT&T Network, is that for a given message a single path is selected and used.

1.5.2.2 Portion of Path Selected

Here we are concerned, not with how many paths are selected, but rather, whether the entire path or a portion of it is selected.

1.5.2.2.1 Entire Path

Many design programs, even for packet switched networks, select entire paths based upon such criteria as length of the paths with maximum excess capacity, and so on [FRANK, 1972], [FRATTA, 1973], [SCHWARTZ, 1975], [SCHWARTZ, 1976], [SEGALL, 1975].

1.5.2.2.2 Proper Subset of a Path

In the analytic model of [GALLAGER, 1976a], [GALLAGER, 1976b] a distributed analysis is given which picks the next link for messages at a node depending upon destination. The method is "optimal" for an ideal observer or when changes in input traffic are slow relative delays for propagating routing information.

1.5.2.2.2.1 Single Link

The Hot Potato, ARPANET and other packet switching schemes correspond to this situation. In the APRANET, a single link is chosen at a time. Similarly, in most circuit switched networks a path is established by sequentially testing links for blocking.

In the case of progressive routing in circuit switching, after a link is selected control is switched to the next node for the next link selection.

1.5.2.3 Restrictions On Path

In many systems, there are very rigid constraints on paths which may be selected.

1.5.2.3.1 Paths Predetermined

In the case of fixed routing, a list of available routes are provided at system initiation, as in the AT&T commercial network. Otherwise, the routing is adaptive. [BROWN, 1975] and [CEGRELL, 1975] consider a scheme halfway between fixed and adaptive, in which modes are changed depending on traffic.

1.5.2.3.2 Path Length Restricted

In AUTOVON [GORGAS, 1968] path lengths have an upper bound, thus restricting number of paths which can be tried in a "home grid."

1.5.2.3.3 Nodes Restricted

An example of node restrictions is hierarchical switching such as in AT&T in which nodes higher in the hierarchy are used only if nodes lower in the hierarchy are occupied [WEBER, 1962].

1.5.2.3.4 No Restrictions

In random routing, [PROSSER, 1962a] any paths may be generated.

1.5.3 How is a Route Terminated?

In all the cases we have discussed, the search for a route is intended to establish communications between a source, NS, of messages and at least one destination, ND, for messages or signals. However, the search may terminate before the destination is included in the route.

1.5.3.1 Last Node in Route is ND

In this case, the message is received at the destination.

1.5.3.1.1 Message Received Incorrectly

In this case, the error check indicates an error in the message.

1.5.3.1.1.1 Route Held

The established route may be held long enough to send a request for retransmission on the same route or even to receive the retransmitted message. In most circuit switching methods, the route is usually held for all retransmissions.

1.5.3.1.1.2 Route Discarded

In a dialup circuit switched system, if there are too many errors, an operator may hang up and establish a new route by redialing. In ARPANET, errors are checked and retransmissions are done at each IMP on a link by link basis; hence, if a message is

received incorrectly at the destination it requires retransmission only on the last link. For a multipacket message, if a packet with a required sequence number does not arrive at ND, then a retransmission is requested from NS, but it may establish a path to ND different from the original one.

1.5.3.1.2 Message Received Correctly

In this case, the error check indicates correct message transmission.

1.5.3.1.2.1 Route Held

In a hybrid system, it is possible to have the packet header carry the signal to establish a route for the circuit switched traffic. Thus, several bits in the header keep track of a route and put a hold on trunks as the packet makes its way through the network. If the packet is received correctly at ND, then the route is held for circuit-switched traffic.

1.5.3.1.2.2 Route Discarded

This is usually the case for message and packet switching.

1.5.3.2 Last Node in Route is Not ND

In this case, the message is not received at the destination and the same types of decisions can be made as in 3.3.1.1 for incorrectly received messages.

1.5.3.2.1 Message is Blocked

In circuit switching a message is blocked if, on each permissible route, at least one link on the node is completely occupied. In this case, the signal never reaches ND to establish a path for the message. In the ARPANET, messages are blocked only if all links or IMP's in a cut set separating NS and ND are down. In this case, a packet does not reach the destination IMP.

1.5.3.2.2 Message is Preempted

In many networks, such as the AT&T telephone network, there are no priorities. In ARPANET there are only two priorities, one for message packets and the other for control packets containing acknowledgements and other protocol information. However, these are not preemptive and hence will not result in elimination of messages and termination of their routes.

In most military systems priorities are present. In AUTOVON several classes of priorities are used [GORGAS, 1968]. In the proposed AUTODIN II network there is a complicated set of 16 level precedences and priorities, some of which are preemptive.

1.5.3.2.3 Message is Discarded

The system itself may discard correct messages and terminate a route based upon built in criteria to prevent excessive utilization of system resources by a message, excessive degradation of a signal, or routing which appears to indicate system malfunction.

1.5.3.2.3.1 Message is Aged

In some schemes priorities are not set by the user or a priori by the system, but rather are based upon the age of the message in

the system and other factors such as traffic or delay [PICKHOLTZ, 1976], [McCOY, 1975]. [BUTRIMENKO, 1972] recommends placing messages at different places in queues depending upon the age of the message. [CEGRELL, 1975] discards messages if their estimated delay in the network exceeds a threshold.

1.5.3.2.3.2 Route Violates Constraints

The system may decide that a constraint on the route length or topology is violated.

1.5.3.2.3.2.1 Path Length

A signal or packet may keep track of how many links have been traversed and may be terminated if the number exceeds a threshold. With digital data which is regenerated at repeaters, the criteria might be inefficient system utilization rather than signal degradation. In some versions of a "ring" network [FARBER, 1972] a packet makes its way around a ring of links and switches until the destination switch accepts the packet or until the packet traverses the ring an excessive number of times, at which point the packet is discarded.

1.5.3.2.3.2.2 Looping

In packet radio, duplicate copies of the same message may be discarded at a repeater. In some systems, a message is discarded if it returns to its original source. In other cases, the message is not allowed to return but is redirected. A special case is "ping-ponging" which refers to loops with two links [NAYLOR, 1976].

REFERENCES

[BARAN, 1964] Baran, P., "On Distributed Communication Networks," IEEE Transactions on Communication Systems, March 1964, pp. 1-9.

[BBN, 1975] BBN Report 3106, 1975.

[BBN, 1969] BBN Report 1783, 1969.

[BHAR, 1974] Bhar, R. and D. C. Majithia, "A Study of Routing Strategies in a Packet Switched Computer Network," University of Waterloo Computer Communications Network Group, Report E-20, May 1974.

[BOEHM, 1964] Boehm, S. P. and P. Baran, "On Distributed Communications II: Digital Simulation of Hot-Potato Routing in a Broadband Distributed Communications Network," Rand Report RM-3103-P4, August 1964.

[BOEHM, 1969] Boehm, B. W. and R. L. Mobley, "Adaptive Routing Techniques for Distributed Communication Systems," IEEE Transactions on Communication Techniques, Vol. COM-17, No. 3, June 1969, pp. 340-349.

[BROWN, 1975] Brown, C. W. and M. Schwartz, "Adaptive Routing in Centralized Computer Communications Networks with an Application to Processor Allocation," IEEE International Conference on Communications (ICC), San Francisco, June 1975, pp. 47-12 - 47-16.

[BUTRIMENKO, 1964] Butrimenko, A. V., "On Searching for the Shortest Paths Along a Graph During Variations in It," Technical Cybernetics (USSR), Vol. 6, 1964.

REFERENCES (Cont'd)

[BUTRIMENKO, 1972] Butrimenko, A. V., "Routing Technique for Message Switching Networks with Message Outdating," Proceedings of the Symposium on Computer-Communication Networks and Teletraffic, Polytechnic Institute of Brooklyn, April 4-6, 1972, pp. 257-261.

[CANTOR, 1974] Cantor, D. G. and M. Gerla, "Optimal Routing in a Packet-Switched Computer Network," IEEE Transactions on Computers, Vol. C-23, No. 10, October 1974, pp. 1062-1069.

[CHOU, 1972] Chou, W., and H. Frank, "Routing Strategies for Computer Network Design," Proceedings of the Symposium on Computer-Communication Networks and Teletraffic, Polytechnic Institute of Brooklyn, April 4-6, 1972, pp. 301-309.

[CEGRELL, 1975] Cegrell, T., "A Routing Procedure for the TIDAS Message-Switching Network," IEEE Transactions on Communications, Vol. COM-23, No. 6, June 1975 pp. 575-585.

[CHYUNG, 1975] Chyung, D.G. and S.M. Reddy, "A Routing Algorithm for Computer Communication Networks," IEEE Transactions on Communications, November 1975, pp. 1371-1373.

[CLAUS, , 1974] Claus, A. and D. J. Kleitman, " Cost Allocation in Networks: The Bulk Supplies Problem," Networks, Vol. 4, No. 1, 1974, pp. 1-18.

REFERENCES (Cont'd)

[FARBER, 1972] Farber, D. J. and K. Larson, "An Experimental Distributed Communication System to Handle Bursty Computer Traffic," Proc. Ass. Comput. Mach. Symp., October 13-16, 1969.

[FISCHER, 1976] Fischer, M. and T. Harris, "A Model for Evaluating the Performance of an Integrated Circuit and Packet-Switched Multiplex Structure, IEEE Transactions on Communications, February 1976, pp. 195-202.

[FRANK, 1971] Frank, H., and W. Chou, "Routing in Computer Networks," Networks, Vol. 1, 1971, pp. 99-112.

[FRANK, 1972] Frank, H., and W. Chou, "Topological Optimization of Computer Networks," Proceedings of IEEE, November 1972, pp. 1385-1397.

[FRANK, 1970] Frank, H., I. T. Frisch, and W. Chou, "Topological Considerations in the Design of the ARPA Computer Network," Spring Joint Computer Conference, Vol. 36, 1970, pp. 581-587.

[FRATTA, 1973] Fratta, L., M. Gerla, and L. Kleinrock, "The Flow Deviation Method: An Approach to Store-and-Forward Communication Network Design," Networks, Vol. 3, No. 2, 1973, pp. 97-133.

[FRISCH, 1975] Frisch, I.T., "Technical Problems in Nationwide Networking and Interconnection," IEEE Transaction on Communication, January, 1975, pp. 78-88.

REFERENCES (Cont'd)

[FULTZ, 1972] Fultz, G. L., "Adaptive Routing Techniques for Message Switching Computer-Communication Networks," School of Engineering and Applied Sciences, University of California, Los Angeles, UCLA-Eng-7252, July 1972.

[FULTZ, 1971] Fultz, G. L., and L. Kleinrock, "Adaptive Routing Techniques for Store-and-Forward Communication Networks," Proceedings of the IEEE International Conference on Communications, 1971.

[GALLAGER, 1976a] Gallager, R. G., "Local Routing Algorithms and Protocols," International Communications Conference (ICC), Philadelphia June 14-16, 1976, pp. 20.17 - 20.20.

[GALLAGER, 1976b] Gallager, R. G., "An Optimal Routing Algorithm Using Distributed Computation," Submitted IEEE Transactions of Communications, 1976.

[GERLA, 1975] Gerla, M., "Deterministic and Adaptive Routing Policies in Packet-Switched Computer Networks," Proc. NTC, 1975, pp. 23-28.

[GERLA, 1973] Gerla, M., W. Chou and H. Frank, "Computational Considerations and Routing Problems for Large Computer Communication Networks," Proceedings of the NTC, 1973, pp.2B-1 - 2B-5.

REFERENCES (Cont'd)

[GITMAN, 1976] Gitman, I., R. M. Van Slyke, and H. Frank, "Routing in Packet Switching Broadcast Radio Networks," IEEE Transactions on Communications, Vol. COM-24, No. 8, August 1976, pp. 926-930.

[GORGAS, 1968] Gorgas, J. W., "The Polygrid Network in AUTOVON," Bell Laboratories Record, July-August 1968, pp. 223-227.

[GOTO, 1973] Goto, S., T. Ohtsuki and T. Yoshimara, "A Shortest Path Calculation Program Based on Code Generation Technique," Proceedings 13th Annual Allerton Conference on Circuit and System Theory, October 1, 1975.

[HARRIS, 1973] Harris, R. J., "Concepts of Optimality in Alternate Routing Networks," 7th International Teletraffic Congress, Stockholm, 1973, pp. 427/1-427/6.

[HSIEH, 1976] Hsieh, W. and A. Kershenbaum, "Constrained Routing in Large Sparse Networks," International Communications Conference (ICC), Philadelphia, June 14-16, 1976.

[HUYNH, 1976] Huynh, D. D., H. Kobayashi, and F. F. Kuo, "Design for Mixed Media Packet Switching Networks," The ALOHA System, University of Hawaii, Technical Report B75-30, December 1975.

REFERENCES (Cont'd)

[HUYNH, 1976] Huynh, D. D., H. Kobayashi, and F. F. Kuo, "Optimal Design of Mixed Media Packet Switching Networks: Routing and Capacity Assignment," Technical Report B76-3, March 1976.

[KATZ, 1967] Katz, S., "Statistical Performance Analysis of a Switched Communications Network," Fifth International Teletraffic Conference, New York, 1967, pp. 566-574.

[KATZ, 1970] Katz, S., "Trunk Engineering of Non-Hierarchical Networks," International Teletraffic Congress, 1970, pp. 142/1-142/8.

[KATZ, 1973] Katz, S., "Long Range Planning of Hierarchical Trunk Networks with Rearrangement Costs," 7th International Teletraffic Congress, 1973, pp. 514/1-514/13.

[KLEINROCK, 1969] Kleinrock, L., "Models For Computer Networks," Proceedings International Conference on Communications, June 1969, pp. 21-9 - 21-15.

[KLEINROCK, 1971] Kleinrock, L. and G. L. Fultz, "Adaptive Routing Techniques for Store-and-Forward Computer-Communication Networks," Computer Networks, International State-of-the-art Report, Infotech, Maidenhead, Berkshire, England, 1971, pp. 541-562.

[KLEINROCK, 1964] Kleinrock, L., Communication Nets: Stochastic Message Flow and Delay, McGraw-Hill, 1964.

REFERENCES (Cont'd)

[KLEINROCK, 1976] Kleinrock, L., Queueing Theory, Vol. II, John Wiley & Sons, Inc., 1976.

[KNEPLEY, 1973] Knepley, J. E., "Minimum Cost Design for Circuit Switched Networks," DCA Technical Note No. 36-73, July 1973.

[KOBAYASHI, 1975] Kobayashi, H. and M. Reiser, "On Generalization of Job Routing Behavior in a Queueing Network Model," IBM, T. J. Watson Research Center, IBM Research Report RC-5679, October 1975.

[LAM, 1976] Lam, S., "Store-and-Forward Buffer Requirements in a Packet Switching Network," IEEE Transactions on Communications, Vol. COM-4, April 1976, pp. 394-403.

[LIN, 1975] Lin, P. M., A. J. Leon, and D. R. Stewart, "Performance and Routing Plan Updating for Networks Employing Originating Office Control," NTC, 1975, pp. 8-1-8-4.

[MAXEMCHUK, 1972] Maxemchuk, N. P., "Dispersity Routing," Ph.D. Dissertation, University of Pennsylvania, 1972.

[McCOY, 1975] McCoy, C., Jr., "Improvements in Routing for Packet-Switched Networks," Ph.D. Dissertation School of Engineering and Applied Sciences, George Washington University, Washington, D. C., 1975.

REFERENCES (Cont'd)

[McGREGOR, 1974] McGregor, P. V., "Load Sharing in a Computer Network," Ph.D. Dissertation, Polytechnic Institute of New York, 1974.

[McQUILLAN, 1974] McQuillan, J., "Adaptive Routing Algorithms for Distributed Computer Networks," Ph.D Dissertation, MIT, BBN Report #2831, May 1974.

[NAC, 1973] Network Analysis Corporation, "The Practical Impact of Recent Computer Advances on the Analysis and Design of Large Scale Networks," ARPA Second Semiannual Technical Report, December 1973.

[NAYLOR, 1975] Naylor, W. E., "A Loop-Free Routing Algorithm for Packet-Switched Networks," Fourth Data Communications Symposium, Quebec City, October 1975.

[PHUC, 1971] Phuc, T., "Traffic Routing Techniques in Telecommunications Networks," International Computer State-of-the-Art Report. C, Infotech, 1971.

[PICKHOLTZ, 1976] Pickholtz, R. L. and C. McCoy, Jr., "Effects of a Priority Discipline on Routing for Packet-Switched Networks," IEEE Transactions on Communications, Vol. COM-24, No. 5, May 1976, pp. 83-89.

REFERENCES (Cont'd)

[PROSSER, 1962a] Prosser, R. T., "Routing Procedures in Communication Networks - Part I: Random Procedures," IEEE Transactions on Communictions, December 1962, pp. 322-324.

[PROSSER, 1962b] Prosser, R. T., "Routing Procedures in Communication Networks - Part II: Directory Procedures," IEEE Transactions on Communications, December 1962, pp. 329-335.

[RUDIN, 1976] Rudin, H., "On Routing and 'Delta Routing': A Taxonomy and Performance Comparison of Techniques for Packet-Switched Networks," IEEE Transactions on Communications, Vol. COM-24, No. 1, January 1976, pp. 43-59.

[SCHWARTZ, 1976] Schwartz, M. and C. K. Cheung, "The Gradient Projection Algorithm for Multiple Routing in Message Switched Networks," IEEE Transactions on Communications, April 1976, pp. 449-456.

[SEGALL, 1975] Segall, A., "New Analytical Models for Dynamic Routing in Computer Networks," National Telecommunications Conference, New Orleans, December 1975, pp. 1-5.

[SILK, 1969] Silk, D. J., "Routing Doctrines and Their Implementation in Message-Switching Networks," Proceedings of the Institution of Electrical Engineering (London), Vol. 116, No. 10, October 1969, p. 1631.

REFERENCES (Cont'd)

[STERN, 1976] Stern, T. E., "A Class of Decentralized Routing Algorithms Using Realization," Columbia University, 1976.

[WEBER, 1962] Weber, J. H., "Some Traffic Characteristics of Communication Networks with Automatic Alternate Routing," BSTJ, Vol. XLI, No. 2, March 1962.

[WILKINSON, 1956] Wilkinson, R. I., "Theories for Toll Traffic Engineering in the U.S.A.," The Bell System Technical Journal, March 1956, pp. 421-514.

APPENDIX A

A TAXONOMY FOR ROUTING METHODS

1. SPEED OF PATH SELECTION, SIGNALS AND MESSAGES

1.1 How Fast is The Routing Method in Establishing a New Path Relative to Message Delay Requirements?

1.1.1 Slow: One shot design

1.1.2 Fast

1.1.2.1 Iterative Design

1.1.2.2 Implementation

1.2 What are the Signal Delays?

1.2.1 Zero

1.2.1.1 Ideal Observer: Design only

1.2.1.2 Isolated

1.2.2 Link Delay

1.2.3 Link & Queueing Delay

1.3 What are The Message Delays Relative to The Signal Delays?

1.3.1 Zero: Packet or message switching

1.3.2 One Link: Reservation packet or message switching

1.3.3 L(NS, ND): Circuit switching

1.3.4 Variable Length: Hybrid

2. MEASUREMENTS

2.1 How Often Are Measurements Made and Routes Updated?

2.1.1 TM = TU = ∞: Deterministic routing

2.1.2 TM, TU Finite: Stochastic routing

2.1.2.1 TM vs TU

2.1.2.1.1 TM < TU

2.1.2.1.2 TM = TU

2.1.3 Uniformity in Space

2.1.3.1 TM(NI) = TM(NJ) for all NI, NJ

2.1.3.2 TM(NI) $\neq$ TM(NJ) for some NI, NJ

2.1.4 Uniformity in Time for Measurement

2.1.4.1 TM(TI) = TM(TJ) for all TI, TJ: Synchronous

2.1.4.2 TM(TI) ≠ TM(TJ) for some TI, TJ

2.1.5 Uniformity in Time for Update

2.1.5.1 TU(TI) = TU(TJ) for all TI, TJ: Synchronous update

2.1.5.2 TU(TI) ≠ TU(TJ) for some TI, TJ: Asynchronous update

2.2 What is Measured?

2.2.1 Outage

2.2.2 Traffic

2.2.3 Delay

2.2.4 Packet or Message Parameters

2.2.4.1 Number of Hops

2.2.4.2 Delay

2.2.4.3 Route

2.2.4.3.1 Partial

2.2.4.3.2 Complete

2.2.4.4 Nothing

2.3 Where Are Measurements Sent To? (i.e., Which Are "Informed Nodes"?)

2.3.1 All Nodes

2.3.2 Proper Subset of Nodes

2.3.2.1 k-th Order Neighbor

2.3.2.1.1 Nearest Neighbor

2.3.2.2 Control Nodes

2.3.2.2.1 One Central Node

2.3.2.2.2 Source, Destination

2.3.2.2.3 Other Set of Control Nodes

2.3.2.3 Nowhere: Isolated

3. DECISIONS

3.1 Where Are Routing Decisions Made?

3.1.1 At Informed Nodes

3.1.2 At a Proper Subset of Informed Nodes

3.1.3 Either Informed Nodes or a Subset

3.2 How is a Route Established?

3.2.1 Number or Paths Selected Simultaneously

3.2.1.1 More Than One Path

3.2.1.1.1 Type of Redundancy

3.2.1.1.1.1 Duplicates Messages Sent Over Different Paths

3.2.1.1.1.2 Different Messages Sent Over Different Paths

3.2.1.1.2 Two Paths

3.2.1.1.2.1 Diverse Paths

3.2.1.1.2.2 Partially Overlapping Paths

3.2.1.2 Single Path

3.2.2 Portion of Path Selected

3.2.2.1 Entire Path

3.2.2.2 Proper Subset of Path

3.2.2.2.1 Single Link

3.2.3 Restrictions on Paths

3.2.3.1 Paths Predetermined

3.2.3.2 Path Length Restricted

3.2.3.3 Nodes Restricted

3.2.3.4 No Restrictions

3.3 How is a Route Terminated

3.3.1 Last Node in Route is ND

3.3.1.1 Message Received Incorrectly

3.3.1.1.1 Route Held

3.3.1.1.2 Route Discarded

3.3.1.2 Message Received Correctly

3.3.1.2.1 Route Held

3.3.1.2.2 Route Discarded

3.3.2 Last Node in Route is Not ND

3.3.2.1 Message is Blocked

3.3.2.2 Message is Preempted

3.3.2.3 Message is Discarded

3.3.2.3.1 Message is Aged

3.3.2.3.2 Route Violates Constraints

3.3.2.3.2.1 Path Length

3.3.2.3.2.2 Looping

APPENDIX B

PREVIOUS CLASSIFICATIONS OF ROUTING PROCEDURES

[PROSSER, 1962a, 1962b]

1. Random

    a. Pure random
    b. First order neighbor
    c. Second order neighbor
    d. k-th order neighbor

2. Directory

    a. Connectivity information
    b. Connectivity plus cost information
    c. Connectivity plus traffic

[BOEHM, 1969]

1. Adaptive

    a. Stochastic

        1. Backwards learning
        2. Negative reinforcement
        3. Biadaptive

    b. Deterministic

        1. Ideal routing

2. Nonadaptive

[FULTZ, 1972]

1. Deterministic

   a. Flooding
   b. Fixed
   c. Network routing and control center
   d. Ideal observer
   e. Split routing

2. Stochastic

   a. Distributed

      1. Asynchronous update
      2. Periodic update

   b. Isolated

      1. Shortest queue plus bias
      2. Local delay estimate

   c. Random

[McQUILLAN, 1974]

1. Control Regime

   a. Deterministic - Fixed control at nodes
   b. Isolated - Independent control at nodes
   c. Distributed - Equal, shared control at nodes
   d. Centralized - Center controls nodes

2. Decision Process

   a. Reachability
   b. Objective function
   c. Traffic assignment

3. Updating Process

   a. Content of routing messages
   b. Propagation of routing messages

4. Forwarding Process

   a. Length of route directory
   b. Width of route directory

[GERLA, 1975]

1. Network Status

   a. Deterministic
   b. Adaptive

2. Locality of Information

   a. Global
   b. Local

3. Routing Computation

   a. Centralized
   b. Distributed

4. Objective

    a. System optimization
    b. User optimization

[RUDIN, 1976]

CENTRALIZED TECHNIQUES

1. Fixed network configuration

2. Network routing control center

3. Fixed between updates

4. Proportional

5. Ideal observer (Present)

6. Ideal observer (Future)

DISTRIBUTED TECHNIQUES

1. Cooperative, periodic or asynchronous update

2. Isolated, local delay estimate

3. Isolated, shortest queue + bias

4. Random

5. Flooding

[BHAR, 1974]

1. Deterministic

   a. Flooding
   b. Fixed routing
   c. Network routing center
   d. Ideal observer

2. Stochastic

   a. Random routing
   b. Isolated - hot potato
   c. Distributed

      1. Synchronous update
      2. Asynchronous update

CHAPTER 2

ANALYSIS OF INTEGRATED SWITCHING LINKS

CHAPTER 2

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CHAPTER 2

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CHAPTER 2

ANALYSIS OF INTEGRATED SWITCHING LINKS

2.1 INTRODUCTION

The analysis of an integrated switching link is addressed in this chapter. The notion of an integrated switching network (of which the link is an element) considered here is one in which traffic can be served on a blocking or delay basis. The integrated switching concept includes elements from circuit switching and elements from packet switching. The switching and transmission facilities of the network are dynamically shared between traffic using the circuit and packet switched modes of operation. That is, the capacity of a link can be dedicated to a circuit switched connection at one instant and carry store-and-forward traffic at another instant. Similarly, the switching node contains all programs and functions needed to perform either circuit switching or packet switching and the central processor (or processors) is shared by all functions; its "instantaneous" load depends on the mix of traffic requirements, the priorities, etc., at the particular time.

The notion of circuit switching in the integrated network may not coincide with the classical notion, in that a physical end-to-end path may not exist while communication takes place in the circuit switched mode, and the traversing of a switch is not entirely transparent. The exact operation of an integrated switching concept would depend on the switching and transmission technologies used to implement it. The circuit switching notion as conceived here is one in which end-to-end switching and transmission facilities are "reserved for" a pair engaged in communication. Information communicated using this mode may be stored and for-

warded when traversing a switching node. In contrast to the packet switched mode, however, the delay of the circuit switched information (during the communication period) when traversing a switch is virtually constant; independent of switch and outgoing link loads. Furthermore, the circuit switched notion as used here is extendable to the case in which one takes advantage of idle (silent) periods in the circuit switched connections to transmit information from other circuit switched connections, signaling messages, or store-and-forward data. The above discussion implies a circuit switched notion which is similar to a "reservation" approach, and in which the network control programs, rather than the end users, manage and control the end-to-end dedicated resources. Hence it is a virtual circuit concept.

Since some switching techniques are more appropriate for a particular type of traffic (eg. [MIYAHARA, 1975]), the most logical form of an integrated switching strategy would incorporate all available methods and judiciously match the "right" switching strategy to a given traffic type; alternatively, the determination of switching mode could be vested with the end user. For example, customers might be willing to tolerate degraded performance using an inappropriate type of switching provided their transmission cost was reduced or due to security/privacy requirements. Independent of the ultimate realization of the integrated switching strategy, the concept proper would offer several advantages. In addition to a reduced need for additional transmission facilities, more efficient channel utilization, consistency of performance for diverse traffic types, novel mixed form of interaction (such as terminal data inquiry with computer voice answer-back or vice-versa) would all be possible. Integrated switching also enables realization of economies of scale resulting from the integration of application categories. Furthermore, each application category includes several message types; including text and protocol and control messages; not all message

types of a category may be best served by the same switching concept. Hence the integrated network concept introduces another degree of freedom which results in the problem of partitioning the message types of all application categories between, say, circuit switching and packet switching modes; or alternatively, developing a set of switching concept protocols which best serve the given application categories. The latter is not addressed in the chapter. The analysis considers an integrated link, the capacity of which is either divided between circuit and packet switched modes of operation or dynamically shared between them. Furthermore, it is implicitly assumed that voice traffic uses the circuit switched mode, and data traffic uses the packet switched mode. When signaling messages used for circuit set up and disconnection are considered they are assumed to use packet switching.

We assume that two streams of traffic which require circuit switching or packet switching service are offered to the integrated link. Analytical models are developed for determining the probability of blocking for the circuit switched traffic and the average delay for the packet switched traffic. The link technology presumed is synchronous time-division multiplexing, and an operation similar to the SENET Concept [COVIELLO, 1975; FISHER, 1976]. A synchronous clock partitions the channel into fixed duration frames; each frame is in turn divided into slots. A boundary is introduced which divides the frame into slots used for circuit switching and slots used for packet switching. The analytical models address the case in which the boundary is fixed, and the so called "movable boundary" where the packet switched traffic can use the excess capacity of slots assigned for circuit switching.

The main objectives for developing the integrated link analysis models were: to enable the design of integrated networks, to study design tradeoffs in such networks, and for comparison of an integrated switching implementation against a packet switching or circuit switching network each carrying both voice and data. This, and the fact that the exact operation of an integrated

link is highly dependent on technology and implementation, placed a requirement of simplicity of the models. Furthermore, it requires the inclusion of specific design parameters in the formulation.

The models developed have the following properties and capabilities:

- Fixed and moveable boundary frame management
- Voice and data priorities and precedence
- Includes parameters of voice digitalization rates and packet size.
- Store-and-forward traffic with two fixed packet sizes.

The experimental results presented in this chapter address the issues of:

- Frame management policies.
- Slot partition strategies between circuit and packet switched traffic.
- Sensitivity of slot assignment and frame management to traffic load and traffic mix.
- Speech interpolation.
- Alternative procedures for incorporating signaling traffic with regular data traffic in the design phase.

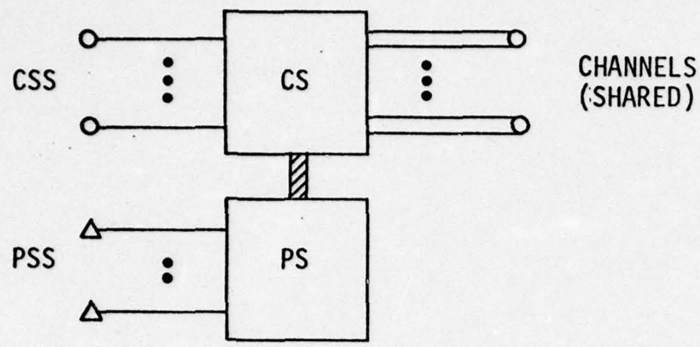
2.2 INTEGRATED SWITCHING NETWORK ALTERNATIVES

Potential integrated switching strategies that could be used in an integrated traffic environment are presented; the strategy being analysed in this chapter is identified and its input parameters detailed. The range of parameters to be used for the experimental studies are also discussed. Distinctions are made between strategies that can be implemented in the near future and those requiring substantial development.

The performance of the most technically viable technique over the long term (synchronous time-division digital multiplexing) is discussed in detail. For all integrated switching strategies, only packet switching is considered to be representative of store-and-forward techniques. Message switching is excluded for convenience, since packet switching will usually provide superior performance except in special cases (e.g. single hop delivery and short message length).

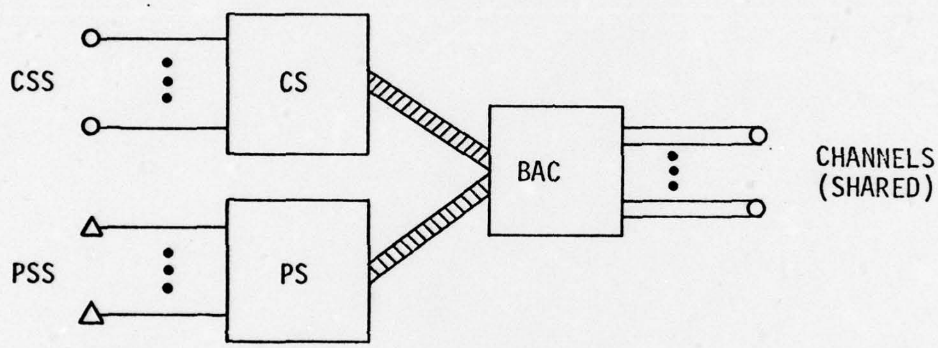
2.2.1 Combination of Existing Techniques

The first strategy illustrated in Figure 1a, assumes that the user community can be apriori partitioned into two distinct classes: those requiring circuit-switched service and those requiring packet switched service. It is conceivable that this distinction may entirely disappear in the future. In order to maximize the use of existing switching and transmission facilities, packet switched service is regarded as simply an adjunct to circuit switching (Strategy 1). Initial access requests for a portion of the transmission bandwidth by packet switched subscribers, are conducted under a circuit super-structure. After appropriate channel capacity has been allocated, however, the actual information transmission is carried out using packet switching techniques. A variation of this concept has been indirectly proposed for use in the Air Force SATIN IV network, which will derive transmission capacity from an existing circuit switched voice network, AUTOVON.

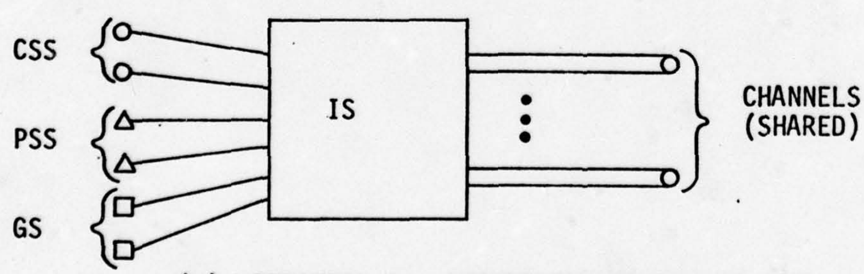


BAC = BANDWIDTH ACCESS CONTROLLER
IS = INTEGRATED SWITCH
CS = CIRCUIT SWITCH
PS = PACKET SWITCH
CSS = CIRCUIT SWITCHED SUBSCRIBERS
PSS = PACKET SWITCHED SUBSCRIBERS
GS = GENERAL TYPE OF SUBSCRIBER

(a) STRATEGY 1: PARTITIONED USER COMMUNITY, CIRCUIT SWITCHED ACCESS TO CHANNELS



(b) STRATEGY 2: PARTITIONED USER COMMUNITY, SHARED ACCESS TO CHANNELS



(c) STRATEGY 3: INTEGRATED USER COMMUNITY, SHARED ACCESS TO CHANNELS

FIGURE 1: POTENTIAL METHODS OF PROVIDING PACKET AND CIRCUIT SWITCHING IN A SINGLE NETWORK

Two consequences of this approach are immediately apparent; if no circuit switched physical path exists between two packet switched subscribers who wish to communicate, a route must first be established before packet transmission can commence. Secondly, if there exists insufficient transmission capacity for a packet switched transmission request, additional capacity must be "dialed-up". One proposed method for determining whether the acquisition of additional channel capacity is necessary is to monitor the packet switch's buffer utilization (i.e. queue length). In the event the queue length exceeds a certain predefined threshold, additional capacity should be acquired. The viability of this approach depends upon the duration of the circuit switch setup procedure. If this becomes too long, the additional protocol delay required for the packet switching flow control-buffer allocate scheme may provide an inferior level of service. As pointed out in [GEBERHARD, 1967], some degree of hysteresis must also be introduced into the packet queue length monitoring and control procedure, in order to avoid unnecessary "chattering" caused by the continual switching of circuits due to statistical fluctuations in packet queue size.

The primary advantage associated with Strategy 1 is the ease of transition into the integrated network environment due to a maximal use of existing facilities.

2.2.2 Longer Term Integrated Switching Strategies

The final two strategies (Figures 1b and 1c) differ according to the types of subscribers that can be interconnected and the overall level of integration which can be achieved. Strategy 2 requires that the subscriber community be partitioned into distinct classes according to their service requirements (packet or circuit switching). Again, existing switching facilities can be used to gain low level network access; however, a bandwidth access controller is additionally needed to control the sharing of transmission capacity between both types of subscribers.

Hence, packet switching transactions establish a channel allocation according to their own unique protocols (as opposed to the circuit switching protocols of Strategy 1). Strategy 3 includes Strategy 2, but also accommodates a third type of "integrated" user, who can transmit information under either switching discipline. Hence, the switch CPU must determine the switching mode to be employed when an integrated subscriber requests transmission capacity. Note that this latter technique may also permit more novel forms of intercommunication (e.g. telephone voice terminal data, etc.) provided the integrated switch of Figure 1 can provide the necessary signal interfaces. Clearly, under Strategy 3 potential savings can accrue due to the shared use of switch hardware/software resources; however, this could be offset by the increased functional complexity and possible resultant operating inefficiency. The existence of the "integrated" subscriber may also imply greater switching flexibility in the network environment; specifically, the reliance on a single switching technique over the end-to-end "connection" is no longer required. Hence, part of the message transmission through the network could be circuit switched, while the remainder is packet switched.

There exist two methods by which the channel capacity can be adaptively shared under either Strategy 2 or 3: frequency division or time-division multiplexing. The first strategy assigns selected portions of the available spectral bandwidth (subchannels) to a specific type of switching. Time division multiplexing employs a synchronous clock which partitions the channel into frames of fixed duration. Each frame is in turn subdivided into slots. The channel allocation process then adaptively assigns a certain subset of the available slots to packet or circuit switched transactions as shown in Figure 2. A "boundary" is introduced which divides the frame into two regions: one supporting packet switching and the other circuit switching. This concept

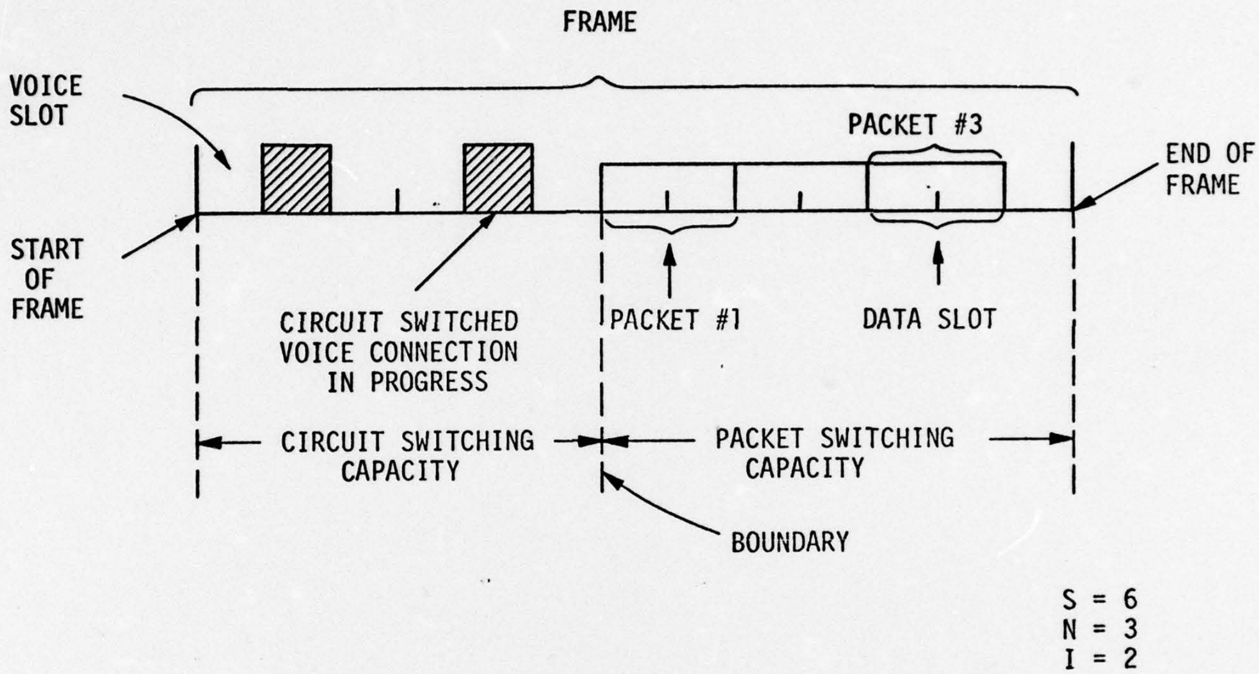


FIGURE 2: TIME-DIVISION MULTIPLEXED INTEGRATED LINE: FRAME STRUCTURE

was initially proposed in [COVIELLO, 1975] and possesses several advantages over FDM. The necessity of guard bands between separate subchannels, intermodulation distortion due to filter nonlinearities and relatively high cost of filters argue heavily against FDM. In addition, TDM readily permits the transmission of a variety of data rates (seizure of multiple time slots per frame) under computer control, whereas FDM does not due to the complex A/D interface with the modulation equipment and the inflexibility of spectral reallocation. Finally, the existence of all digital synchronous transmission facilities will greatly simplify the implementation of the outlined hybrid switching strategy. This capability currently exists in several countries (e.g., Bell's DDS in the U.S.). Strategy 3 and specifically the channel architecture in Figure 2 is being modelled in this chapter.

2.2.3 Input Parameters For The Integrated Link Model

A list of all input parameters to the analytic models characterizing the performance of the integrated switching system is shown in Figure 3. For the present investigation, we have limited the parameter range to a select representative set of values as now described.

The specific integration scenario that will be used for experimentation envisions the combination of AUTOVON (circuit voice network) and AUTODIN II (packet data network) into a single integrated network that supports an integrated switching strategy. Limited traffic information describing each facility exists; however, projections for the average busy hour are available as shown in Figure 4. All information regarding data traffic has been excerpted from [DCA, 1975]; all information concerning voice traffic can be found in [DCA, 1975] and [GTE, 1975].

A. Traffic Parameters:

1. Voice
   a. Mean Call Origination Rate = λ call/sec.
   b. Average Call Duration = h seconds
   c. Voice Digitization Rate = VDR bits per seconds per voice channel
   d. Speech Activity Factor = y percentage of time speaker is active
2. Data
   a. Mean Packet Arrival Rate = θ packets/sec.
   b. Fixed Packet Length = P bits
3. Priorities (2 priority traffic classes are used with each switching method)
   a. Percentage of High and Low Priority Voice
   b. Percentage of High and Low Priority Data

B. Channel Parameters:

1. Inherent Characteristics
   a. Channel Capacity = C bps
   b. Frame Duration (Inverse Clock Rate) = b seconds
2. Allocation and Operation
   a. Number of Voice Slots per frame = S slots
   b. Data-to-Voice Slot Ration = I
   c. Number of Data Slots per frame = N slots
   d. Slot Allocation Policy (Fixed or Moving Boundary)
   e. Precedence Handling (Priority based or Traffic based)

C. Performance Parameters:

1. Circuit Switched Voice Blocking Probability = B
2. Packet Switched Data Average Delay = D

FIGURE 3: RELEVANT MODEL PARAMETERS

TIME FRAME	CONUS AUTOVON[1]	AUTODIN II[2](BITS/HR)	AUTODIN II[3](BITS/SEC)
1978	2184E	7.2×10^9	8.1×10^6
1985	2510E	14.4×10^9	10×10^7

(a) TOTAL BACKBONE NETWORK TRAFFIC FOR BUSY HOUR

TIME FRAME	VOICE[4]	AVERAGE DATA[5] (bps)	PEAK DATA[5] (bps)
1978	36.40E	$.25 \times 10^6$	1.0125×10^6
1985	41.80E	$.5 \times 10^6$	1.25×10^6

(b) REPRESENTATIVE NODAL TRAFFIC

NOTES:

1. Traffic in erlangs, represents 20% of total originated traffic and assumes a 2% annual voice growth rate. 80% of the total offered traffic is intra-switch (traffic originated and destined for subscribers attached to the same switch).
2. Average busy hour (51.5% of traffic is intra-switch).
3. Peak busy second in busy hour.
4. Assumes 60 node CONUS AUTOVON topology (uniform traffic distribution).
5. Assumes 8 node AUTODIN II topology (uniform traffic distribution). (See Figure 4 for network topology).

FIGURE 4: GLOBAL TRAFFIC VOLUME PROJECTIONS

From Figure 4b, with h=1 minute, call arrival rates varying from 25 calls/minute to 50 calls/minute adequately characterize the CONUS AUTOVON nodal circuit switch voice load. The 1978 average packet data throughput estimate and 1985 peak throughput estimate shown in Figure 4b, serve as effective upper and lower bounds for the AUTODIN II nodal traffic. For the purpose of analysis, a packet length P=1000 bits is postulated. Although in actuality, the traffic distribution is non-uniform, we will assume that the packet arrival rate varies from 250 packets/sec. to 1250 packets/sec. $(250 < \theta < 1250)$.

A diversity of voice digitization techniques exist. The projected use of such techniques in the overall Defense Communication System (circa 1985) is given in Figure 5 [GTE, 1975]. Over the long term, a phased transition towards a heavier reliance on the advanced techniques such as APC, LPC and vocoding may take place. The current experiments will account for the use of all techniques, bounded at the extremes by conventional PCM at VDR=64 kbps and vocoder methods at VDR=2.4 Kbps. A rudimentary analysis concerning potential bandwidth savings, when a speech interpolation mechanism is incorporated for circuit switched voice, will also be conducted as a function of the speech activity factor, y (25-75%).

The percentage of high priority traffic (both voice and data) will be parametrically varied between the extremes of 1% (normal) and 30% (critical). In addition, two precedence orderings among the traffic types will be studied. Precedence as a function of traffic type and precedence as a function of priority class.

DIGITIZATION METHOD	DIGITIZATION RATE (Kbps)	% USE IN THE DCS (CIRCA 1985)
PCM - Pulse Code Modulation	48 - 64	5
DPCM - Differential Pulse Code Modulation	32 - 48	10
CVSD - Continuous Variable Slope Delta Modulation	16 - 32	50
APC - Adaptive Predictive Coding	8 - 16	15
LPC - Linear Predictive Coding	4 - 8	10
VOCODERS	2.4 - 8	10

FIGURE 5: VOICE DIGITIZATION TECHNIQUES

2.3 MODELS FOR INTEGRATED LINK ANALYSIS

2.3.1 Integrated Link Operation

Consider the frame structure shown in Figure 2. A channel of capacity C bits per second is assumed to be clocked synchronously at fixed intervals of b seconds. The periodic clocking "breaks up" the channel capacity into frames of duration b seconds. The frame is further partitioned into several "slots" of fixed or variable duration. Each slot is sized to accomodate a single circuit switched voice connection. Since, voice is isochronously generated at a rate of VDR bps, every frame (b seconds), exactly X bits must be provided (voice slot size):

$$X = b \cdot VDR \tag{1}$$

and the duration of a voice slot therefore becomes

$$d = \frac{X}{C} \tag{2}$$

The length of a data packet P is assumed to be an integral number of voice slots I wide:

$$P = IX \tag{3}$$

hence, the packet transmission time is given by:

$$t = Id \tag{4}$$

It will henceforth be referred to as the voice-to-data slot ratio (i.e.,a single data slot (packet) requires I voice slots for transmission).

During each frame, a certain amount of the total available slot capacity is reserved exclusively for circuit switched voice transmission. The number of circuit switched slots is denoted

as S. The remainder of the individual voice slots are available for packet data transmission, S-NI. Given the number of dedicated circuit switched voice slots, S, the number of data packet slots N can be derived as:

$$N = \left[\frac{\frac{b}{d} - S}{I}\right] \qquad (5)$$

where [F] represents the greatest integer less than or equal to F. The actual position of the data and voice slots within the frame does not influence the blocking probability analysis, but will determine the complexity of the integrated switch operation. Generally, the isochronously generated voice could be bit, character or block interleaved; if bit interleaving is not used, some form of voice buffer storage must be provided (elastic stores). The size of the storage will depend on the frame duration and voice digitization rate. For convenience, we assume no periodic interleaving within the frame, so that all slots of a particular traffic type (voice or data) are "packed" contiguously within a frame as in Figure 2. Hence, elastic storage must be provided for the incoming voice (since it is transmitted only at periodic intervals). Additionally, the elastic storage is used to compensate for any phase or frequency jitter which could potentially exist in the incoming digital bit stream.

As shown in Figure 2, there exists a boundary separating a circuit switched voice region and a packet switched data region. An incoming circuit switched voice call request, which arrives at a certain instant relative to the beginning of the frame, is buffered for at most one frame period in a "gating" queue; in the interim, the switch CPU ascertains whether an idle circuit switched voice slot is available which can be assigned to the call. If one exists, the call reserves the available slot and is permitted to use it, the next frame. If no slots are currently idle, the call is "blocked" and subsequently flushed from the gating queue. Incoming data packets are buffered and wait in

queue indefinitely until a data slot becomes available for transmission. The data packet queueing delay could, therefore, extend over several frames. In actuality, the switch operation/interrupt structure may require a formal scheduling of packet transmission; however, for the purposes of analysis, packet data transmission is assumed to be asynchronous.

As shown in Figure 6, there exist many potential slot allocation policies, depending on the number of boundaries, size of the shared channel capacity and boundary "hardness". A fixed boundary policy (Policy 1) allocates a certain invariant portion of the slots to each traffic type. In the event, idle slot capacity exists in either region, it cannot be used by the other traffic type. The moveable boundary policy gives inherent priority to circuit switched voice, by allowing it to use up to a maximum of S slots per frame. However, if there exists a sufficient number of idle voice slots during a given frame, Policy 2 allows packet data to utilize this currently available slot capacity. Policy 3 provides no formal boundary and allows either traffic type to compete for the fully shared slot capacity on a first-come first-served or priority basis. This strategy has disadvantages since a particular traffic type could temporarily exclude other traffic classes from access to the channel and does not realistically distinguish between buffered (data) and nonbuffered traffic (voice). The final strategy is a combination of Policies 1 and 3, and provides dedicated voice and data slot allocations, in addition to a mutually shared region. The analysis of Policies 3 and 4 is quite complex, and therefore attention throughout this work will be focused exclusively on Policies 1 and 2.

2.3.2 Analytical Models For An Integrated Link

An exact analytical model for determining the probability of blocking and the average delay was developed in [FISHER, 1976]. This model was not adopted for integrated network design due to its computational complexity and because it does not explicitly

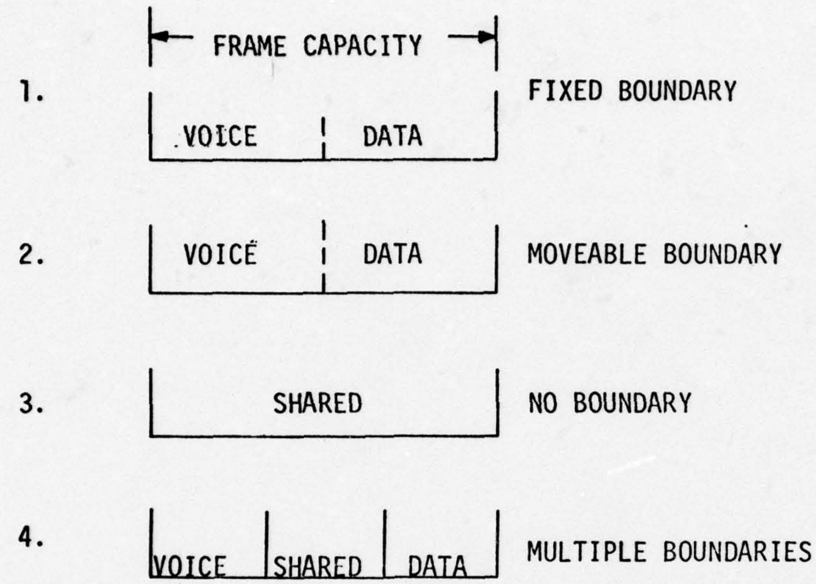


FIGURE 6: TIME-DIVISION MULTIPLEXED INTEGRATED LINK: POSSIBLE CHANNEL ALLOCATION POLICIES

incorporate the packet size as a parameter. Furthermore, it is not easily extendable to circuit-switched and packed-switched priority traffic.

The computational complexity involves the solution of the roots of a transcendental equation representing the generating function of the steady state probabilities of having a certain number of packets in the system at the start of a frame. An inaccuracy is introduced into the analysis in [FISHER, 1976] by examining the data slot occupancy status only at discrete time intervals equal to the frame duration. Although this enables an inherently single server queueing system to be represented as a multiserver system, the packet transmission time is implicitly equated to the frame duration. Hence, the actual packet length P (bits) is not considered in the formulation. The end result of this assumption is that those packets transmitted using a certain number of slots per frame are considered to be buffered until the end of the frame, when in actuality they depart from the system as soon as their transmission is completed.

Alternative models are now proposed which can prove advantageous in more readily obtaining performance measures. The analytic simplicity afforded by the alternative formulations is also attractive in that it enables a quantitative examination of other more important issues to be conducted (e.g., channel allocation policies, priority traffic classes, non-Poisson input traffic and different packet sizes). The ultimate benefits that can be attributed to these models are: a relative simplicity of formulation; closed form solution and thus, increased computational efficiency; straightforward adaptation for use in an integrated network design procedure.

Two models for integrated link analysis are presented. Both approximate the circuit switched blocking probability by the Erlang B formula, and differ in the model for evaluating the average packet delay.

2.3.2.1 Blocking Probability Approximation For Circuit-Switched Traffic

In this section we compare the Erlang B approximation with the exact values in [FISHER, 1976] and show its sufficiency for the practical range of integrated link operation. It is assumed that calls originate according to a Poisson distribution with mean rate λ, the holding time is exponential with mean $h=1/\mu$ and the number of voice slots in the integrated link is S.

The blocking probability is approximated by:

$$B \cong B(a,S) = \frac{\frac{a^S}{S!}}{\sum_{i=0}^{S} \frac{a^i}{i!}} \qquad (6)$$

where $a = \lambda h = \lambda/\mu$ is the offered circuit switched voice load in Erlangs.

Figure 7 shows the "exact" blocking probability and the Erlang B approximation as a function of the frame duration b, for $\lambda = 10$, $S = 10$; for fixed B, independent of the frame duration. One can see that there exists a certain threshold value of frame size beyond which as the frame duration increases, the blocking probability increases quite rapidly. Over a certain range, ($b \leq .1$ sec.), the actual integrated switched channel blocking performance and that predicted by the Erlang B equation are virtually indistinguishable.

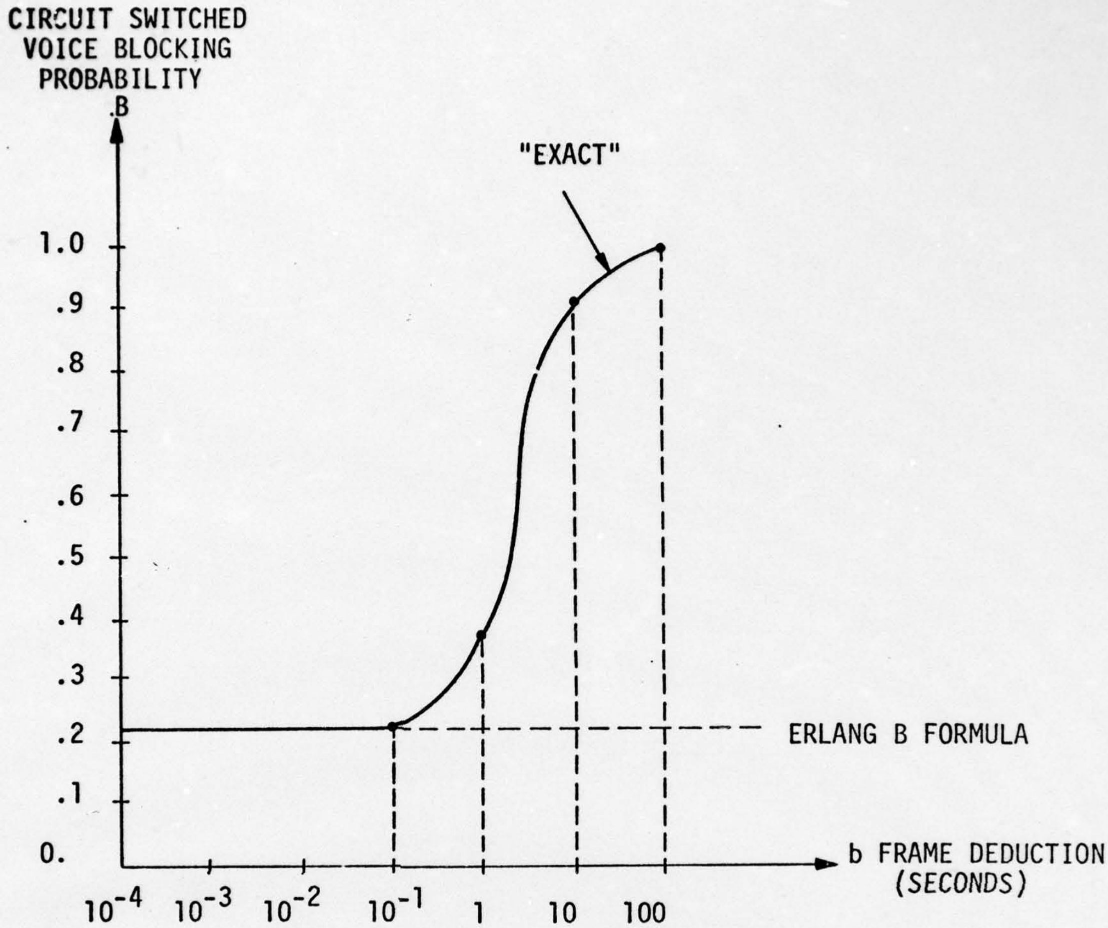
An analytical comparison is now provided for the special case S = 1. The results according to [FISHER, 1976] are:

$$\Pi_0 = \frac{1-e^{-\mu b}}{1-e^{-\mu b}e^{-\lambda b}} \qquad (7)$$

and

$$\Pi_1 = \frac{e^{-\mu b}\,(1-e^{-\lambda b})}{1-e^{-\mu b}\,e^{\lambda b}} \qquad (8)$$

where Π_i is the steady state probability of i calls in the system.



NUMBER OF VOICE SLOTS = S = 10

MEAN CALL ARRIVAL RATE = λ = 10 CALLS/SECOND

MEAN CALL HOLDING TIME = $\frac{1}{\mu}$ = h = 1 SECOND

ERLANG B IS INVARIANT WITH RESPECT TO FRAME DURATION

FIGURE 7: BLOCKING PROBABILITY DEPENDENCE UPON FRAME DURATION FOR THE HYBRID SWITCHING CHANNEL

Using the Erlang B equation, the probability that j of a total of S slots are occupied with an input traffic intensity of a erlangs is given by:

$$\Pi(a,j,S) = \frac{\frac{a^j}{j!}}{\sum_{L=0}^{S} \frac{a^L}{L!}} \tag{9}$$

hence, for S=1:

$$\Pi(a,0,1) = \frac{1}{1+a} \tag{10}$$

and

$$\Pi(a,1,1) = \frac{a}{1+a} \tag{11}$$

For the Erlang B equation to provide an adequate approximation to the channel blocking probability performance, the following equalities should hold:

$$\left.\begin{aligned} \Pi_0 &= \Pi(a,0,1) \\ \Pi_1 &= \Pi(a,1,1) \end{aligned}\right\} \tag{12}$$

As a direct consequence of Eq. (12), the offered voice load a, input to the integrated channel can be expressed (omitting algebraic details) as:

$$a = \frac{1-e^{-\lambda b}}{e^{\mu b}-1} \tag{13}$$

or equivalently:

$$a = \frac{\lambda b - \frac{(\lambda b)^2}{2!} + \frac{(\lambda b)^3}{3!} - \cdots}{\mu b + \frac{(\mu b)^2}{2!} + \frac{(\mu b)^3}{3!} + \cdots} \tag{14}$$

Hence, for an offered load a, the Erlang B approximation is appropriate only if λb (mean number of call requests per frame) and μb (mean number of call disconnects per frame) is small, so that the higher order terms of both numerator and denominator in Eq. (14) can be neglected. For the general integrated switching

channel with S circuit switched slots, the ratios $\frac{\lambda b}{S}$ and $\frac{\mu b}{S}$ representing the average number of call originations and disconnections per slot must be small, $\frac{\lambda b}{S} << 1$, $\frac{\mu b}{S} << 1$.

An intuitive explanation for the blocking dependence on frame duration is now given. The channel is only available for voice traffic at periodic intervals, incoming calls that attempt to seize a circuit switched slot are forced to wait until the start of the frame. Furthermore, calls which disconnect do not relinquish their slot capacity "immediately", since the "freed-up" voice slot does not become available to new calls until the next frame. Since, pending call requests can wait in the "gating" queue for at most one frame period, as the frame duration increases, more requests will queue per frame. Assuming there exists a fixed number S of circuit switched voice slots (fixed VDR and C), as the frame duration increases, a higher percentage of queued call requests will be blocked. In short, the frame introduces a "clustering" among the incoming call requests by allowing calls to accumulate and seize available channel capacity only at selected intervals; thus, the effective call request stream appears to be more irregular than Poisson and will encounter higher blocking.

Throughout the remainder of the analysis, we will assume that the Erlang B formula Eq. (6) is a sufficiently accurate approximation to the actual circuit switching portion of channel performance, i.e.,

$$\Pi_i = \frac{\frac{a^i}{i!}}{\sum_{j=0}^{S} \frac{a^j}{j!}} \qquad (15)$$

Under the Erlang B assumption of Eq. (15), the average voice slot utilization ε is given by:

$$\varepsilon = a\,(1-B(a,S)) \qquad (16)$$

As we have analytically demonstrated, the applicability of the Erlang B approximation becomes worse as the frame duration

increases. However, there are physical factors related to the actual switch operation which prevent the frame itself from becoming too long. Specifically, the frame duration is restricted to a specific region of size:

$$b_{min} < b < b_{max} \tag{17}$$

If the frame duration becomes shorter than b_{min}, the required slot management activities that must be performed periodically each frame, begin to pose an excessively high processing burden on the switch CPU. Furthermore, short frames provide inefficient channel utilization due to the excessive amount of overhead represented by the frame control header. Apart from the unacceptably high level of blocking, other physical factors prohibit the frame from becoming too long ($>b_{max}$). As the frame duration increases, the required buffer capacity for the temporary storage of incoming circuit switched and packet switched traffic becomes excessive. This storage capacity can be subdivided into three classes: elastic storage for ongoing digitized voice; the "gating" queue used for scheduling of incoming circuit switched call requests; and store-and-forward buffers to hold data packet arrivals pending channel availability. Finally, circuit-switched call requests must wait a period of time equal to b/2 on the average, prior to receiving an allocation or reservation of trunk capacity. However, depending on the routing/signaling employed, the set-up delay constraints governing circuit switched calls could require an appreciably shorter frame.

2.3.2.2 A Simple Approximation For Packet Switched Traffic

Before presenting our models for the packet-switched traffic, an approximation to the model in [FISHER, 1976] is given which enables utilizing known results from queueing theory and a closed form solution. In [FISHER, 1976], the packet switched traffic is essentially modelled by an M/D/N queueing system to which

a "common" approximation is an M/M/N system. Invoking known results from queueing theory for the latter model, the average queueing delay for a K channel multi-server queue is given by:

$$W(\theta,b,K) = \left(\frac{(\theta b)^K}{(K-1)!} \frac{\frac{1}{b}}{(K/b-\theta)^2} \right) \gamma \qquad (18)$$

where

$$\gamma = \frac{1}{\sum_{m=0}^{K-1} \frac{1}{m!} (\theta b)^m + \frac{1}{K!} (\theta b)^K \left(\frac{K}{K-\theta b}\right)} \qquad (19)$$

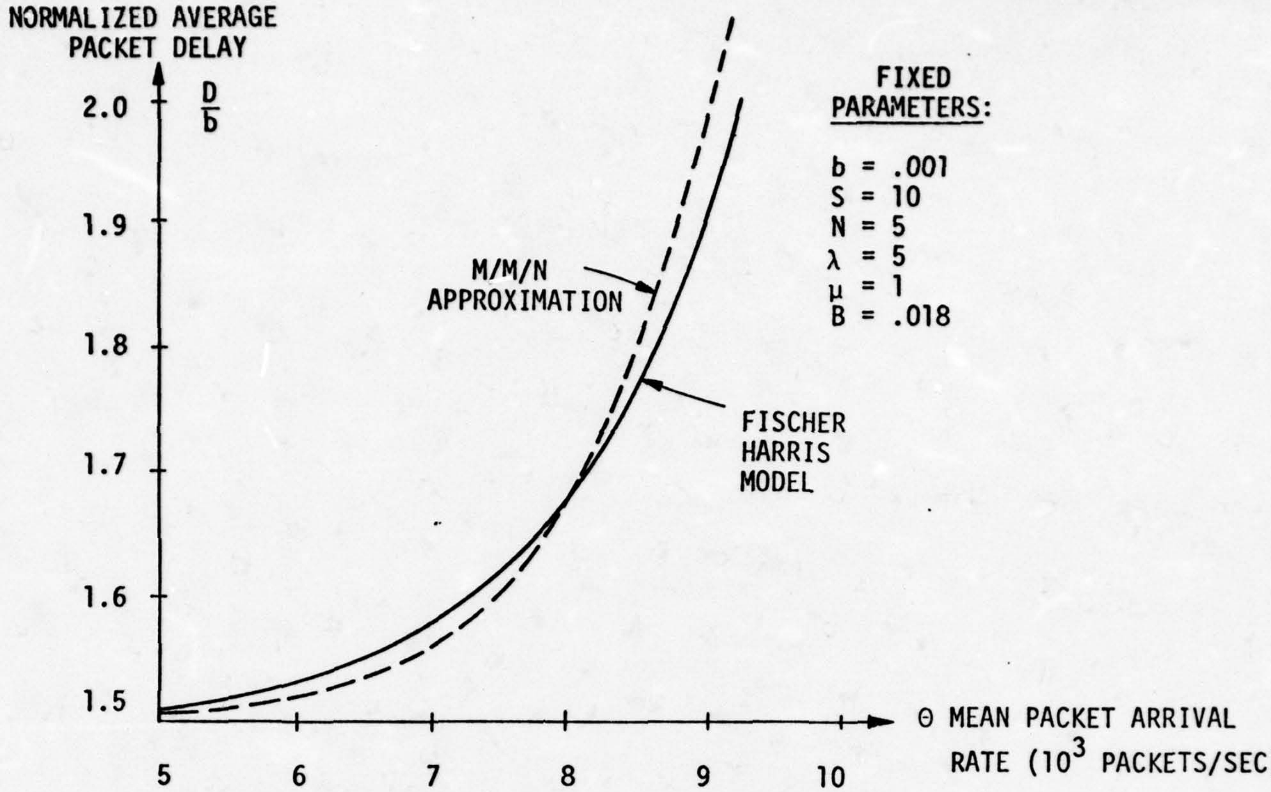
In the above, b (the frame duration) is taken to be the average service time required to transmit a single packet over the channel. Hence, this approximation also neglects the actual packet length. The average packet delay (queueing and transmission) can now be expressed as:

$$D = \frac{b}{2} + b + W(\theta,b,K) \qquad (20)$$

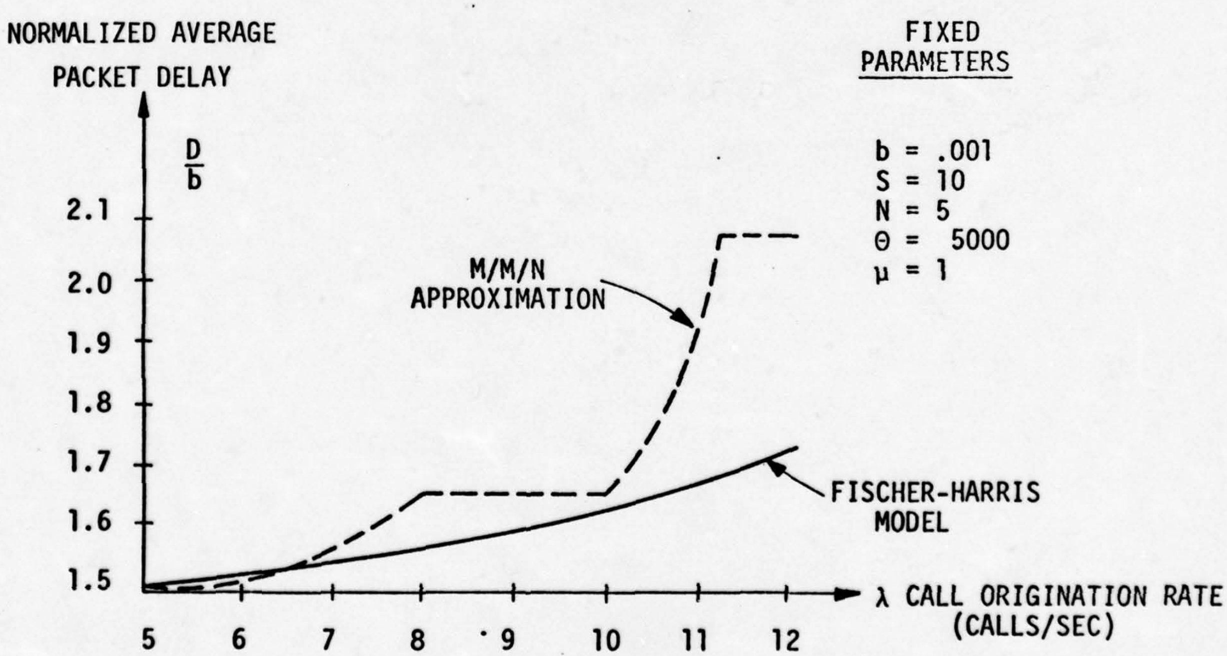
where

$$K = \begin{cases} N, & \text{fixed boundary} \\ N + \left[\frac{S-\varepsilon}{I}\right], & \text{movable boundary} \end{cases} \qquad (21)$$

with ε as in Eq. (16). The second part in Eq. (21) represents the average number of data slots in the moveable boundary case, where the second term is due to the excess voice slot capacity. In Eq. (20), the first two terms represent average delay due to the framing and the pseudo-transmission time = b. Due to the assumed exponential nature of the service (packet transmission) time, Eq. (20) should overestimate the mean packet delay. However, the presence of the greatest integer function [•] in Eq. (21) inhibits the approximation from providing the desired bounding property as illustrated in Figure 8. An attempt was made to incorporate the actual packet length into this approximation; however, this resulted in virtually no queueing delay, using the parameter values of Figure 8, due to the markedly reduced service time.



(a) AVERAGE PACKET DELAY UNDER MOVING BOUNDARY SLOT ALLOCATION POLICY



(b) AVERAGE PACKET DELAY UNDER MOVING BOUNDARY POLICY AS A FUNCTION OF VOICE LOAD

FIGURE 8: COMPARISON BETWEEN FISCHER-HARRIS FORMULATION AND APPROXIMATION MODELS

All models considered up to this point have implicitly regarded the transmission of buffered data packets as an inherently parallel process via the use of multiserver queueing systems. Thus, the formulation assumes that the packets can be transmitted over several slots simultaneously. This is clearly not indicative of the actual integrated switching channel's operation. In the following section, we derive the packet data performance for single server based models of the integrated switching channel.

2.3.3 Single Server Models For Packet Switched Traffic (An Alternative Formulation)

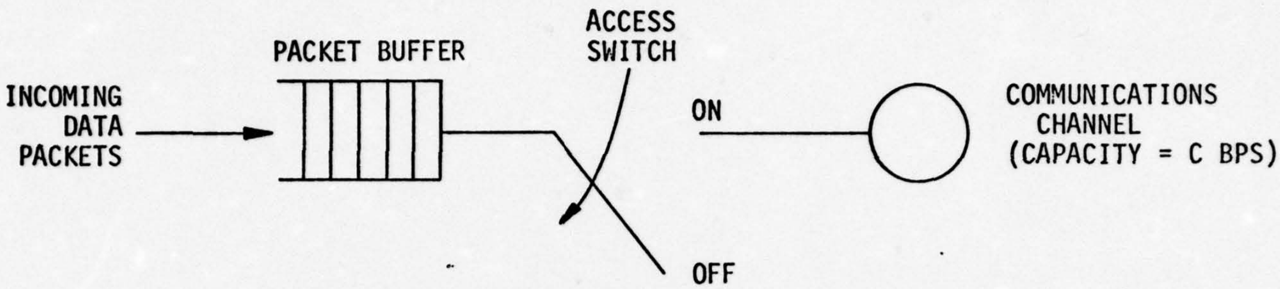
Consider the integrated switched communication channel from the perspective of packet data traffic as depicted in Figure 9. Incoming data packets arrive at the switch, are buffered and await transmission access to the channel. The channel is available for packet transmission only for a portion of the entire frame on a periodic basis. The duration of the channel access depends on the slot allocation policy. The "throwing" of the channel access switch is due to the presence of the circuit switched voice at the start of each frame.

For the sake of simplicity we assume that the channel is intermittently available for packet transmission each frame, and that a stochastic decision regarding the potential repositioning of the channel access switch is made at discrete temporal intervals equal to the packet transmission time, t. Specifically, let σ be the probability that the channel is currently available for the transmission of a packet. The value of σ will depend on the slot allocation policy employed. In particular:

$$\sigma = \sigma_F = \frac{N}{N+[S/I]} \tag{22}$$

for the fixed boundary data slot allocation policy, and

$$\sigma = \sigma_M = \frac{N+[\frac{S-\varepsilon}{I}]}{N+[S/I]} \tag{23}$$



PACKETS ARRIVE ACCORDING TO A POISSON PROCESS (MEAN ARRIVAL RATE = Θ PACKETS/SEC.)

PACKETS ARE FIXED LENGTH = P BITS

PACKET TRANSMISSION TIME = t = P/C

IN ACTUALITY, SWITCH IS "OFF" FOR THE CIRCUIT VOICE PORTION OF THE FRAME, SWITCH IS "ON" FOR THE PACKET DATA PORTION OF THE FRAME

PROBABILITY SWITCH IS "ON" = σ

PROBABILITY SWITCH IS "OFF" = $1-\sigma$

FIGURE 9: SINGLE SERVER MODEL FOR THE CHANNEL UNDER AN INTEGRATED SWITCHING DISCIPLINE

for the moving boundary data slot allocation policy, with ϵ defined in Eq. (16) is the average voice slot occupancy. The above equations represent the percentage of data slots available for packet transmission per frame.

Based on Eqs. (22) and (23) and the model in Figure 9, we now formulate the state equations describing the operation of the integrated channel. Let p_i represent the probability that i packets are present in the system (on the channel and buffered) at the start of the current data slot. Denote by γ_K the probability that K "fresh" packets arrive to the system during a data slot interval = t. Note that no explicit statement regarding the packet arrival process (Poisson or otherwise) has been made or implied by γ_K. A discussion of the applicability of this formulation for a general arrival process is given in the Appendix A. Then:

$$p_i = \sum_{K=0}^{i-1} \gamma_K \quad (1-\sigma)p_{i-K} + \sigma p_{i+1-K} + \gamma_i(p_0+\sigma p_1) \qquad (24)$$

which indicates that the possibility of i packets in the system at the start of the current data slot depends on the number of packets in the system at the beginning of the previous data slot, the number of subsequently arriving "fresh" packets and the position of the channel access switch during the last data slot. The generating function for the state probabilities p_i is given by (see Appendix A):

$$P(z) = \frac{\sigma p_0 (z-1)}{z(\sigma + \frac{1}{\Omega(z)} - 1) - \sigma} \qquad (25)$$

where:

$$p_0 = 1 - \frac{\theta t}{\sigma} \qquad (26)$$

and $\Omega(z)$ is the probability generating function for the packet arrival process:

$$\Omega(z) = \sum_{j=0}^{\infty} \gamma_j \; z^j \qquad (27)$$

Using the probability generating function $P(z)$, the average number of packets L in the integrating switching system can be obtained from:

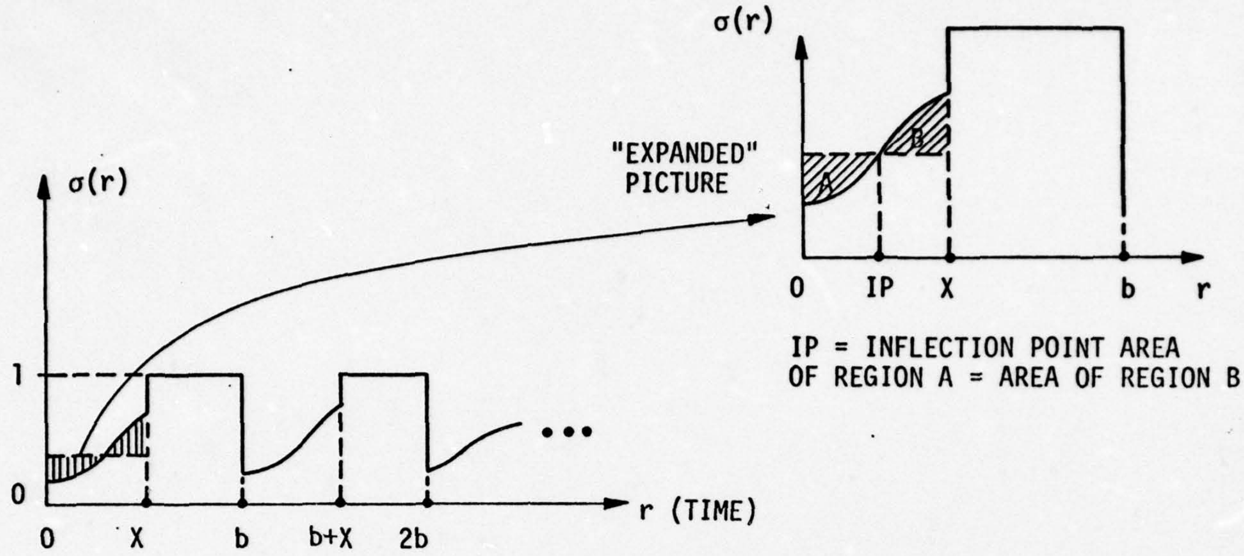
$$L = \left. \frac{\partial P(z)}{\partial z} \right|_{z=1} \tag{28}$$

whereupon an application of Little's Formula ($L=\Theta D$), finally yields the average packet delay. If the packet arrival process is assumed to be Poisson, the average packet delay becomes:

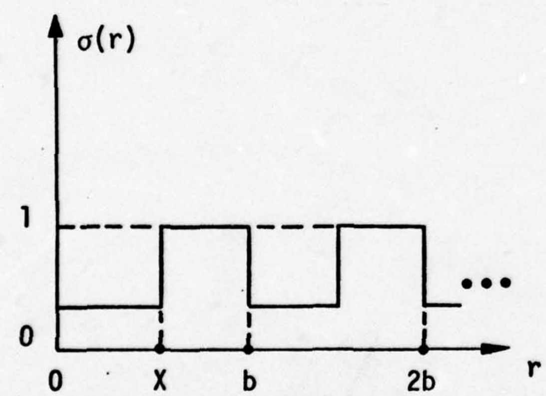
$$D = \frac{t(2-\Theta t)}{2(\sigma-\Theta t} \tag{29}$$

Other packet arrival processes can be studied by appropriate substitution into Eq. (25) and subsequent differentiation as in Eq.(28). We refer to this model as the "partial availability model". Note that for $\sigma= 1$, Eq.(29) reduces to the well known M/D/1 queueing system, since the channel possesses full availability (switch is always "on").

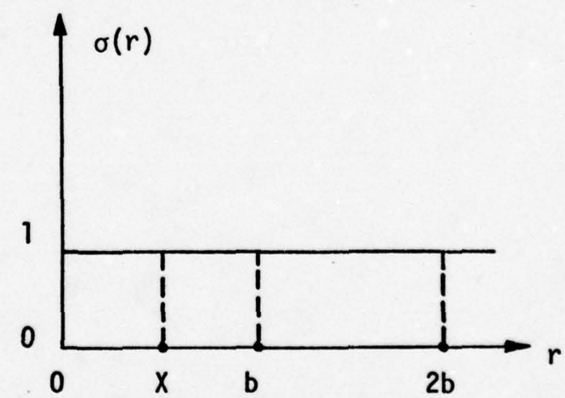
This last observation suggests a far less formal way by which to obtain an equivalent result for the average packet delay. Consider the channel availability time profile $\sigma(r)$ shown in Figure 10. The function $\sigma(r)$ represents the probability that at time r, the channel is available for packet data transmission. In steady state, this function is periodic at integral values of the frame duration for the integrated switching channel. For the first frame, under the moving boundary slot allocation policy, the channel is available with certain probability strictly less than one for the voice portion of the frame, $r < x$. For $r > x$, (the data portion of the frame), the channel is completely available $\sigma(r) = 1$. Over the course of the voice portion of the frame $\sigma(r)$ is montonically increasing due to the order of the voice slot seizure (i.e., the first available slots in the frame are seized). The functional behavior of $\sigma(r)$ for $r < x$ will change depending on the order of idle voice slot seizure.



(a) CHANNEL AVAILABILITY TIME PROFILE FOR MOVING BOUNDARY SLOT ALLOCATION POLICY



(b) FIRST ORDER APPROXIMATION



(c) ZERO ORDER APPROXIMATION

FIGURE 10: CHANNEL AVAILABILITY PROFILE

Several approximations can be made to $\sigma(r)$ as illustrated in Figure 10b and 10c. The so called zero order approximation $\sigma(r) =$ constant over all time, is indicative of the analysis performed earlier. Regardless of the actual form of $\sigma(r)$, as long as it is periodic, there exists an "effective" per frame channel capacity C* given by:

$$C^* = C \left(\frac{1}{b} \int_0^b \sigma(r)\, dr \right) \tag{30}$$

Due to the inherent "averaging" performed by the integration in Eq. (30), the value of the integral C* is independent of the nature of the $\sigma(r)$ approximation used. In our system, C* can be obtained directly as:

$$C^* = C \left(\frac{NId}{b}\right) \tag{31}$$

for the fixed boundary policy, and

$$C^* = C \; \frac{(NI+S-\varepsilon)d}{b} \tag{32}$$

for the moving boundary policy with d equal to the voice slot duration.

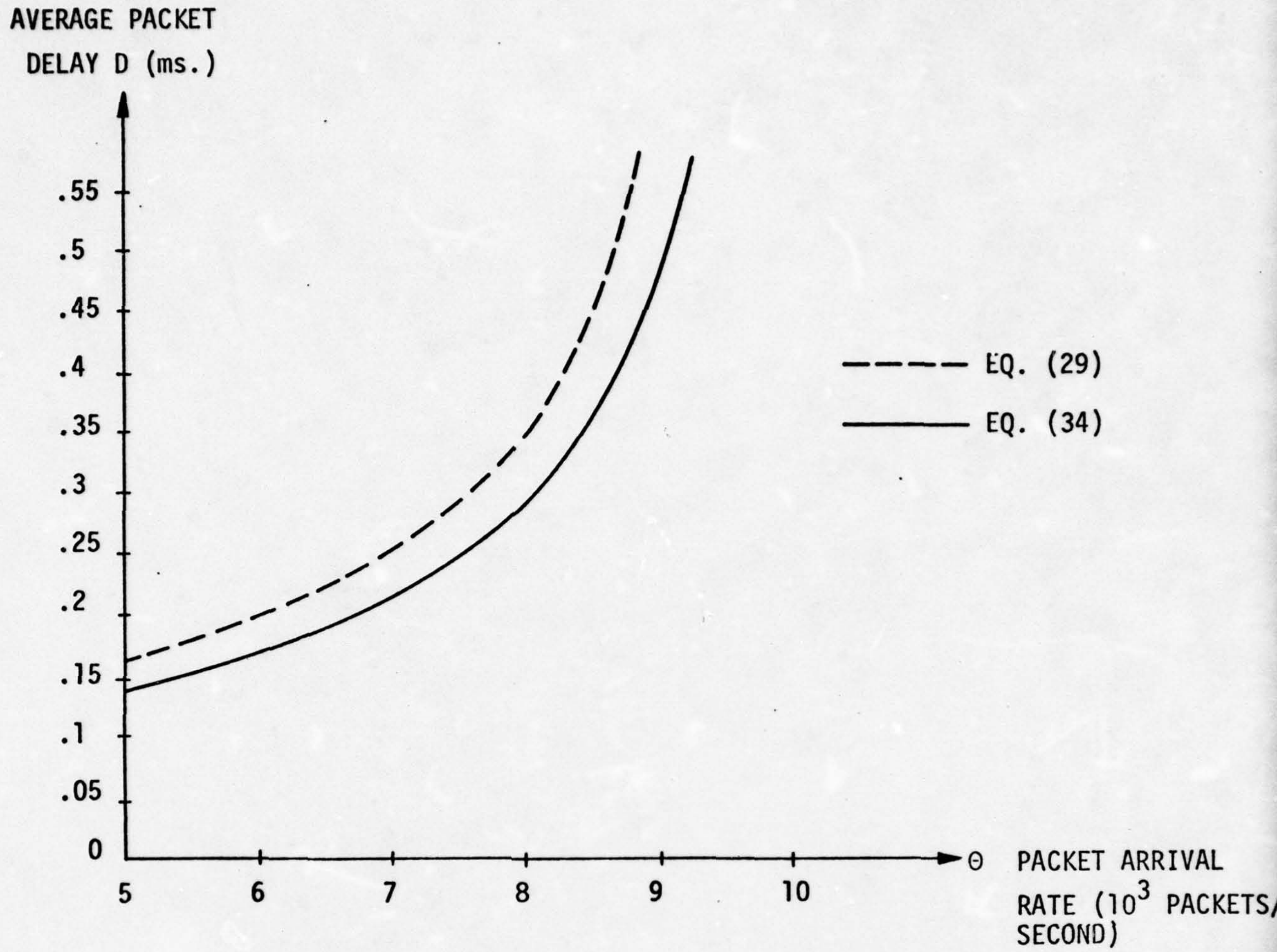
An "effective" packet transmission time t* can then be derived as:

$$t^* = {}^{P}/_{C^*} \tag{33}$$

Finally, the average data packet delay, can be derived as the mean delay in an M/D/1 queueing system with service time equal to the "effective" packet transmission time, t* or:

$$D = t^* + \frac{\theta(t^*)^2}{2(1-\theta t^*)} = \frac{t^*(2-\theta t^*)}{2(1-\theta t^*)} \tag{34}$$

We refer to this model as the "effective capacity model". Observe that Eq. (34) bears a striking functional resemblance to Eq. (29). A comparison between the different formulations Eq. (29) (dashed) and Eq. (34) (solid) is shown in Figure 11. Agreement is acceptable throughout the parameter range illustrated.



PARAMETERS:

C = CHANNEL CAPACITY = 1.5 Mbps
b = FRAME DURATION = 1 ms.
VDR = VOICE DIGITIZATION RATE = 100 Kbps
P = PACKET LENGTH = 100 BITS
S = NUMBER OF VOICE SLOTS PER FRAME = 10
N = NUMBER OF DATA SLOTS PER FRAME = 5
λ = CALL ARRIVAL RATE= 5 CALLS/SEC.
h = CALL HOLDING TIME= 1 SEC.

FIGURE 11: AVERAGE PACKET DELAY FOR MOVING BOUNDARY SLOT ALLOCATION POLICY

2.4 ANALYTIC EXTENSIONS OF THE PERFORMANCE MODELS

The relative simplicty of the models developed in the preceding section enables a more detailed investigation of relevant issues. The issues addressed in the section are: impact on channel bandwidth efficiency of speech interpolation, influence of several types of circuit switched traffic on overall performance, and design issues regarding the operation of an integrated channel in a priority driven environment.

2.4.1 Speech Interpolation

A considerable increase in the efficiency of channel bandwidth utilization can be gained if a speech interpolation capability for the circuit-switched voice slots is provided. For a circuit switched voice connection, once the appropriate voice slots have been seized, they are held for the duration of the conversation. However, during the course of the ongoing digitized speech, there are several temporal intervals during which no voice is present (idle periods). This phenomenon arises for several reasons; voice is inherently bursty and consists of a sequence of speech and idle periods; the more sophisticated voice digitization methods may not detect sufficient changes in the speech waveform to generate an additional sample; or finally, the speaker may be momentarily pausing in order to await a verbal response from the listener. In any case, there are periods during which the already seized but temporarily idle voice slots could be better utilized to service additional voice connection requests or packetized data.

The implementation of the speech interpolation capability is often quite complex. In brief, a speech activity detection mechanism must first ascertain when a speaker is active or silent. The latency associated with the determination that the transition (speaker active, speaker silent) has occurred is referred to as the "hangover" period: Clearly, if the hangover period is considerably longer than the frame duration b, and the speech

idle period is much shorter than the frame duration, a speech interpolation capability does not afford any increases in the efficiency of bandwidth utilization. Even if speech interpolation offers benefits, the frame control is substantially more complex, since the integrated switch at the "other" end must know which voice channels are currently active. Because of the adaptive nature of the sharing among voice slots (no permanently dedicated capacity), "freezeouts" (clipping) of ongoing speech can also occur as a result of the temporary unavailability of slots. Interim buffering of "frozen" speech or a priority rule among active conversations may then be required.

Assuming that the above implementation difficulties have been overcome, we now analyze the potential impact of speech interpolation. Let us define the speech activity factor y as the probability that a circuit switched voice connection is active (speaker is talking). Observe that the use of a speech activity factor is an oversimplified representation of the conversation in that it does not account for the actual temporal nature of the speech (talk spurts and idle periods). The probability that a circuit switched voice connection request is blocked under a speech interpolation discipline is:

$$\Pr(\text{Blocking}) = \Pr\,(S \text{ active connections}) \tag{35}$$

which is the probability that S active circuit switched connections exist over the current frame, independent of the number of total connections. This is given explicitly as:

$$\Pr(\text{blocking}) = \frac{\dfrac{(ay)^S}{S!}}{\displaystyle\sum_{i=0}^{S} \frac{(ay)^i}{i!}} = B(ay,S) \tag{36}$$

where a is the original offered voice load, y the activity factor, and S is the number of voice slots per frame. Similarly the probability that there exist j active voice connections is simply (see Appendix B):

$$\Pi(ay,j,S) \tag{37}$$

Finally, the average voice slot utilization when speech interpolation is provided is given as:

$$\epsilon_{SI} = ay\ (1-B(ay,S)) \tag{38}$$

The remainder of the preceding analyses (average data packet delay) can thus be obtained using ϵ_{SI} in Eq.(38) for Eqs.(21) and (32). Since y 1, speech interpolation in general yields an increase in the effective available voice slot bandwidth. Finally, it should be emphasized that this analysis can be applied to any intermittent form of transmission that is circuit switched such as an iteractive terminal's data transaction (y is a measure of burstiness).

2.4.2 Multiple Voice Traffic Classes

The Erlang B Blocking Formula is applicable regardless of the slot holding time distribution (possible non-exponential), as long as connection requests originate according to a Poisson process. Moreover, if there are L types of circuit switched voice calls possessing different general holding time distributions, but which arrive to the switch according to a Poisson process, the probability that an incoming call is blocked (regardless of type) is given by:

$$\Pr(\text{Blocking}) = B\ (a,S) \tag{39}$$

where a is the cumulative offerred load over all traffic types i.e.,

$$a = \sum_{i=0}^{L} a_i \tag{40}$$

and again, the average voice slot utilization is given as:

$$\epsilon = a\ (1-B\ (a,S)) \tag{41}$$

If the call arrival process is non-Poisson but successive arrivals are independent, the call origination process is referred to as a renewal process. For an exponentially distributed call duration, closed form results are available under renewal input (due to Tackacs (see [RIORDAN, 1962])). Specifically, the probability that an incoming call is blocked is given as:

$$B^* (a,S) = \Pr \text{ (call is blocked)} = \frac{1}{\sum_{j=0}^{S} \binom{S}{j} \frac{1}{x_j}} \tag{42}$$

where:

$$x_j = \prod_{i=1}^{j} \frac{\delta(i/h)}{1-\delta(i/h)} \tag{43}$$

h is the mean call duration, and

$$\delta(s) = \int_0^\infty e^{-st} \, dA(t) \tag{44}$$

is the Laplace-Stieljes Transform of the call arrival process, A(t). For non-Poisson input, the probability an incoming call is blocked (arriving calls distribution) is not equivalent to the probability that all voice slots are busy (external observer's distribution). The probability that all slots are occupied is given as (see Eq. 42):

$$\tilde{B} (a,S) = \frac{aB^*(a,S-1)}{S} \tag{45}$$

The average voice slot utilization also simply becomes:

$$\epsilon = a\,(1-\tilde{B}\,(a,S)) \tag{46}$$

2.4.3 Multiple Data Traffic Classes With And Without Priorities

2.4.3.1 Introduction

In this section we investigate the average packet delays when two classes of packet traffic with or without priorities

and with possible different packet size are offered to an integrated link. The motivation for this investigation is the accommodation of regular packet data traffic and signaling messages (used for circuit set up and disconnection) on the data portion of the integrated link.

We now state the specific assumptions governing the link model considered in this section:

1. Two classes of packet input traffic (class 1 and class 2) with independent Poisson distribution arrivals of rates θ_1 and θ_2 respectively.

2. Each class has a constant packet size P_i, $i = 1, 2$.

3. Channel capacity is C.

4. The partial availability model for packet transmission is assumed, i.e., the packet at the head of the packet queue has probability σ of being transmitted, where σ is given by either Eq.(22) or Eq.(23). If the transmission is unsuccessful, then a packet of class i has to wait P_i/C seconds (i.e., the packet transmission time) for another trail.

5. Regardless of class, once a packet is being scanned for transmission, it will always be the first one transmitted whenever the channel is free.

Assumption 5 implies the following: Suppose a packet has just been transmitted, and suppose the new head-of-the-line packet is of low priority (i.e., at the time of the packet departure, either there is only low priority packets in the system, or there are no packets in the system and the first packet arrival is of low priority). Then this low priority packet is always the first one transmitted whenever the channel is free, regardless of whether there are high priority packet arrivals during the period of channel's unavailability.

2.4.3.2 The Priority Case

We assume that the class 1 packets have priority over the class 2 packets in service, except as noted in Assumption 5. The priority discipline is head-of-the line (HOL) and non-preemptive.

Under Assumptions 4 and 5 the service time for a class i packet is simply the sum of the time spent during scanning until the channel is free, and the transmission time. The moments of the service time distribution for a class i packet can be evaluated as follows: For i = 1,2 let S_i be the service time for a class i packet, and $t_i = P_i/C$ the packet transmission time for a class i packet or the scanning interval for a class i packet then, for n = 1, 2,.......,

$$E(S_i^n) = \sum_{k=0}^{\infty} (kt_i+t_i)^n \times Pr \left\{ \begin{matrix} \text{first successful scanning} \\ \text{occurs at the } (k+1)^{-th} \text{trial} \end{matrix} \right\}$$

$$= \sum_{k=0}^{\infty} (k+1)^n t_i^n (1-\sigma)^k \sigma$$

$$= t_i^n \sum_{k=0}^{\infty} (k+1)^n (1-\sigma)^k. \tag{47}$$

Note that for any positive real number a with $|a|<1$,

$$\sum_{k=0}^{\infty} (k+1)\, a^k = \frac{1}{(1-a)^2}, \tag{48}$$

$$\sum_{k=0}^{\infty} (k+1)^2 a^k = \frac{1+a}{(1-a)^3}. \tag{49}$$

Consequently, the first two moments are given by;

$$E(S_i) = t_i \sigma \frac{1}{\sigma^2} = \frac{t_i}{\sigma} \tag{50}$$

$$E(S_i^2) = t_i^2 \sigma \frac{2-\sigma}{\sigma^3} \doteq \left(\frac{t_i}{\sigma}\right)^2 (2-\sigma) \tag{51}$$

In [MILLER, 1960] it is shown that in a non-preemptive HOL priority queueing system with two classes of traffic, each with a (possibly different) Poisson arrival distribution, and a (possibly different) general service time distribution the mean waiting times W_1 and W_2 are given by;

$$\left.\begin{aligned} W_1 &= \frac{\theta_1 E(S_1^{\,2}) + \theta_2\, E(S_2^2)}{2(1-\theta_1\, E(S_1)\,)}\,, \\ W_2 &= \frac{\theta_1 E(S_1^{\,2}) + \theta_2\, E(S_2^2)}{2(1-\theta_1 E(S_1)\,)\,(1-\theta_1 E(S_1) - \theta_2 E(S_2)\,)} \end{aligned}\right\} \tag{52}$$

and the mean time in the system, T_1 and T_2, are given by

$$T_i = W_i + E(S_i),\ i = 1,2, \tag{53}$$

where θ_i are the class i mean arrival rate, and $E(S_i)$ and $E(S_i^{\,2})$ are the first and the second moments of the class i service time distribution, i=1,2. Consequently, for our model, the mean waiting times are given by;

$$W_1 = \frac{(\theta_1 t_1^{\,2} + \theta_2 t_2^{\,2})\quad(2-\sigma)}{2\sigma(\sigma-\theta_1 t_1)} \tag{54}$$

$$W_2 = \frac{(\theta_1 t_1^{\,2} + \theta_2 t_2^{\,2})\quad(2-\sigma)}{2\ (\sigma-\theta_1 t_1)\ (\sigma-\theta_1 t_1 - \theta_2 t_2)}\,,$$

and the mean delays are given by

$$T_i = W_i + \frac{t_i}{\sigma}\,,\quad i = 1,\ 2. \tag{55}$$

Note that if $\theta_2 = 0$, then

$$T_1 = \frac{\theta_1 t_1^{\,2}(2-\sigma)}{2\sigma(\sigma-\theta_1 t_1)} + \frac{t_1}{\sigma} = \frac{t_1(2-\theta_1 t_1)}{2(\sigma-\theta_1 t_1)}\,, \tag{56}$$

which reduces to Eq. (29). We can express the mean waiting times and the mean delays in terms of flows and capacity (which is

more expressive in the context of network design). Let $C^*(=\sigma C)$ be the effective capacity of the link for packet switched traffic, and $f_i(=\theta_i P_i)$ be the average class i flow in the link. Using C^* and f_i, Equations (54) and (55) can be written as:

$$\left.\begin{aligned} W_1 &= \frac{f_1P_1+f_2P_2}{2C^*(C^*-f_1)}(2-\sigma), \\ W_2 &= \frac{f_1P_1+f_2P_2}{2(C^*-f_1)(C^*-f_1-f_2)}(2-\sigma); \end{aligned}\right\} \tag{57}$$

$$T_i = W_i + \frac{P_i}{C^*},\ i = 1,2. \tag{58}$$

We now consider a generalization of the packet link queueing model in which both the probability of transmission and the scanning interval are allowed to be different for different packet classes. Specifically, we modify Assumption 4 to be as follows:

<u>Assumption 4':</u> Suppose the packet at the head of the queue is of class i, i = 1, 2. Then with probability σ_i the channel is free, and the packet can be transmitted. σ_i is assumed constant regardless of the number of retrials. If the channel is not free, then the channel will be scanned again in t_i seconds.

Evidently, in order for the above model to be a reasonable approximation of packet service in the integrated link, the σ_i's and the t_i's cannot be independent. The σ_i's and the t_i's are related to one another through the average probability that the channel is free and the packet transmission time for two classes.

With Assumption 4', the moments of the packet service time distributions are given by;

$$E(S_i^n) = \sum_{k=0}^{\infty}\left[kt_i + \begin{pmatrix}\text{packet transmission time}\\ \text{for a class i packet}\end{pmatrix}\right]^n \times P_r\begin{Bmatrix}\text{first successful scanning}\\ \text{occurs at the }(k+1)^{-th}\text{ trial}\end{Bmatrix}$$

$$= \sum_{k=0}^{\infty}(kt_1 + \frac{P_i}{C})^n \times (1-\sigma_i)^k\sigma_i \tag{59}$$

Consequently, for $i = 1, 2$, the first two moments are;

$$E(S_i) = \frac{t_i}{\sigma_i}(1-\sigma_i) + \frac{P_i}{C} \tag{60}$$

$$E(S_i^2) = \left(\frac{t_i}{\sigma_i}\right)^2 (1-\sigma_i) + \left[\frac{P_i}{C} + \frac{t_i}{\sigma_i}(1-\sigma_i)\right]^2 \tag{61}$$

With the first two moments of the service distributions determined, the mean queueing time and the mean delay can be obtained from Eqs. (52) and (53).

2.4.3.3 The Non-Priority Case

In this section, we assume that the packets are served on a first-come-first served basis without priority. Following the development in [ANCKER, 1961], we can consider the packets as forming one composite class, with the arrival distribution being the composite of the two arrival distributions, and the service distribution being the composite of the two service time distributions.

Since the arrival processes for both classes are assumed Poisson, the arrival process for the composite is also Poisson, and the mean arrival rate θ is given by:

$$\theta = \theta_1 + \theta_2 \tag{62}$$

The composite service time distribution is given by

$$S(t) = \sum_{i=1,2} S_i(t) \; P_r \begin{Bmatrix} \text{the packet demanding} \\ \text{service is of class } i \end{Bmatrix} \tag{63}$$

where $S_i(t)$ is the density function of the class i. Since the packet arrivals are independent and exponentially distributed

P_r {the packet demanding service is of class i}

$$= P_r \text{ \{the packet next to arrive in queue is of class i\}}$$

$$= \int_0^\infty \theta_i e^{-\theta_i x} \left[\int_x^\infty \theta_{2-i} e^{-\theta_{2-i} y} dy\right] dx \tag{64}$$

$$= \int_0^\infty \theta_i e^{-\theta_i x} e^{-\theta_{2-i} x} dx = \int_0^\infty \theta_i e^{-\theta x} dx = \theta_i/\theta.$$

Consequently,

$$S(t) = \frac{1}{\theta} (\theta_1 S_1(t) + \theta_2 S_2(t)). \tag{65}$$

From which it follows that for $n = 1, 2, \ldots\ldots\ldots,$

$$E(S^n) = \frac{1}{\theta} (\theta_1 E(S_1{}^n) + \theta_2 E(S_2{}^n)) \tag{66}$$

Substituting Eqs. (50) and (51) into the above, we obtain

$$E(S) = \frac{1}{\theta\sigma} (\theta_1 t_1 + \theta_2 t_2), \tag{67}$$

$$E(S^2) = \frac{2-\sigma}{\theta\sigma^2} (\theta_1 t_1{}^2 + \theta_2 t_2{}^2) \tag{68}$$

Now, according to Polloczek-Khintchine [SAATY, 1961], the average queueing time for a process with Poisson arrival distribution and general service time distribution is given by:

$$W_q = \frac{\theta E(S^2)}{2(1-\theta E(S))}, \tag{69}$$

where θ is the mean arrival rate, and $E(S)$ and $E(S^2)$ are the first and the second moments of the service time distribution. Substituting Eqs. (67) and (68) into Eq. (69), we obtain;

$$W_q = \frac{(\theta_1 t_1{}^2 + \theta_2 t_2{}^2)\ (2-\sigma)}{2\sigma(\sigma - \lambda_1 t_1 - \lambda_2 t_2)} . \tag{70}$$

The average delay (queueing plus service) for the two classes are;

$$T_i = W_q + \frac{t_i}{\sigma}, \ i = 1, 2. \tag{71}$$

We can express W_q and the T_i 's in terms of the flows and the effective capacity:

$$W_q = \frac{f_1 P_1 + f_2 P_2}{2C^*(C^* - f_1 - f_2)} \ (2-\sigma), \tag{72}$$

$$T_i = W_q + \frac{P_i}{C^*}, \ i = 1, 2. \tag{73}$$

2.4.4 Voice and Data Priority Classes and Precedence Ordering

In the realistic command and control environment, multiple priority classes may exist. Although extensive analyses have been conducted for multiple priority classes belonging to a single traffic type, few results are available when more than one type of traffic is present. This is the problem that must be addressed in the fully integrated traffic environment (heterogeneous traffic types, multiple priority classes).

For the sake of tractability, we restrict the current analysis to the case of two types of traffic (circuit switched voice and packet switched data) and two priority classes (high (priority 1) and low (priority 2)). The performance measures obtained are as before: circuit switched voice blocking probability and packet switched data average delay. Among the more interesting issues we investigate is the impact of the precedence ordering among traffic types and priority classes. In particular, the two precedence orderings depicted in Figure 12 are analyzed. As indicated, there exists sufficient motivation for examining both precedence orderings. It is appealing to argue that voice is always of higher precedence, since the impact of slot unavailability on voice is far more severe (traffic lost) than on data (longer wait), and delivery of voice must typically occur in real time. However, critical instances could conceivably arise

VOICE 1 VOICE 2 DATA 1 DATA 2 ↓	ORDER OF DECREASING PRECEDENCE	NOTE: The numerals 1 and 2 denote high priority and low priority, respectively

Precedence Motivation: All voice is of inherently high precedence, since data is buffered but voice is lost (blocked)

(a) PRECEDENCE ORDERING 1 (Based on Traffic Type)

VOICE 1 DATA 1 VOICE 2 DATA 2 ↓	ORDER OF DECREASING PRECEDENCE

Precedence Motivation: High priority traffic is of inherently higher precedence than low priority traffic. Within priority class, voice is of inherently higher precedence than data due to the buffering capability given to data.

(b) PRECEDENCE ORDERING 2 (Based on Priority Class)

FIGURE 12: PRECEDENCE ORDERINGS AMONG TRAFFIC TYPES AND PRIORITY CLASSES

where high priority packet data will have greater importance than low priority circuit switched voice. In either case, it is reasonable to assume that high priority voice always has the highest precedence and that low priority data has the lowest precedence. The operational details of the priority mechanism are outlined in greater detail in conjunction with the analysis of both precedence orderings. In an effort to obtain closed form results, several simplifying assumptions are employed regarding class interaction.

2.4.4.1 Precedence Ordering 1 (Based on Traffic Type)

The identical statistical assumptions (Poisson arrival, exponential call holding times) and symbolic notation hold as before. Let λ_i represent the call origination rate for circuit switched voice of priority i, i=1,2. Similarly, Θ_i denotes the packet arrival rate for data priority class i,. i=1, 2. The voice offered load (in Erlangs) of priority i is:

$$a_i = \lambda_i h \tag{74}$$

where h is the mean call duration for both priority classes. Let the variables $B_{i,j}$ and $D_{i,j}$ represent the voice blocking probability and average data delay for priority i traffic under precedence ordering j. The blocking probability for priority 1 voice under precedence ordering 1 is as given as:

$$B_{1,1} = B(a_1, S) = \frac{\frac{a_1^S}{S!}}{\sum_{i=0}^{S} \frac{a_1^i}{i!}} \tag{75}$$

Under precedence ordering 1, all voice is given access to the S dedicated voice slots with potential data sharing in the event idle voice slot capacity exists. Furthermore, it is assumed that high priority voice can preempt low priority voice traffic,

in which case the lower priority voice is subsequently lost. The blocking for priority 2 voice under precedence ordering 1 is thus equal to:

$$B_{2,1} = B(a_1+a_2,S) = \frac{\dfrac{(a_1+a_2)^S}{S!}}{\displaystyle\sum_{i=0}^{S} \frac{(a_1+a_2)^i}{i!}} \tag{76}$$

The above measure accounts for the event that a low priority call is blocked upon arrival. An equally important measure relative to the low priority voice is the probability that a call in progress is preempted (PP).

$$PP = \frac{\lambda_1}{\lambda_1 + \lambda_2} (B_{2,1}-B_{1,1}) \tag{77}$$

An explanation for the above equation is now provided. An incoming high priority call can only preempt an ongoing low priority call if all voice slots are occupied, and at least one low priority call is in progress. The probability that all circuit switching slots are busy is given as $B_{2,1}$ in Eq. (76); excluding the possibility that the busy condition is due exclusively to high priority traffic (probability of occurrence, $B_{1,1}$(Equation (76)), the probability that an incoming high priority call preempts a low priority call becomes simply: $B_{2,1}-B_{1,1}$. The probability that a low priority connection is broken due to an incoming call (which is of higher priority with probability $\lambda_1/(\lambda_1+ \lambda_2)$) is given by PP. Hence, the average voice slot occupancy under precedence ordering 1 becomes:

$$\varepsilon_1 = a_1(1-B_{1,1}) + a_2(1-B_{2,1}-PP) \tag{78}$$

The average data packet delay analysis now presented closely parallels that which led to Eq. (29). All packets (high and low priority) are buffered. Each priority class possesses the

same packet length and the priority discipline between classes is non-preemptive. Given the average voice slot utilization, the average data packet delay for high priority data can be expressed as:

$$D_{1,1} = \frac{t}{2} \frac{(2-\Theta_1 t)}{(\delta-\Theta_1 t)} \tag{79}$$

with

$$\delta = \frac{N}{N + [S/I]} \tag{80}$$

for the fixed boundary slot allocation policy and,

$$\delta = \frac{N + \left[\frac{S-\epsilon_1}{I}\right]}{N+[S/I]} \tag{81}$$

for the moving boundary slot allocation policy, with N the number of data slots, Θ_1 the high priority average packet arrival rate, and t the packet transmission time.

For a single server queue with Poisson arrivals and an arbitrary service distribution, the average queueing delay for the K^{th} priority class in a nonpreemptive operating environment with L total priority classes was derived in [COBHAM, 1954] as:

$$W_K = \frac{\beta(1+\gamma^2)t}{(1- \sum_{i=1}^{K} \beta_i)\ (1- \sum_{i=1}^{K-1} \beta_i)} \tag{82}$$

where β_i is the offered data load (traffic intensity) for priority class i, t is the average service time, γ^2 is the coefficient of variation for the service time distribution and β is the total offered data load ($\beta = \sum_{i=1}^{L} \beta_i$). For the M/D/1 queue with two priority classes, the average queueing delay for priority class 2 reduces from Eq. (82) to:

$$W_2 = \frac{(\beta_1+\beta_2)t}{(1-\beta_1)\ (1-(\beta_1+\beta_2))} \tag{83}$$

A general relation can be derived from Eq.(83), that expresses the average data queueing delay experienced by priority 2 traffic

as a function of the priority 1 traffic delay. In particular:

$$W_2 = \frac{W_1}{(1-(\beta_1+\beta_2))} \tag{84}$$

Employing the concept of "effective capacity model" we make the following straightforward extension from Eq. (79), to obtain the average total packet delay for priority 2 data traffic as:

$$D_{2,1} = \frac{D_{1,1}-t}{(\hat{\delta}-(\theta_1+\theta_2)t)} + t = \frac{t(\theta_1 t+2(1-\hat{\delta})}{2(\hat{\delta}-\theta_1 t)\ (\hat{\delta}-(\theta_1+\theta_2)t)} + t \tag{86}$$

for both fixed and moving boundary slot allocation policies.

2.4.4.2 <u>Precedence Ordering 2 (Based on Priority Class)</u>

The performance measures associated with each traffic type and priority class are now determined relative to precedence ordering 2 (Figure 12b). Due to the complexity of the interaction between traffic types, only an approximate analysis for the low priority traffic is possible.

As in precedence ordering 1, high priority voice is unaffected by the presence of the other traffic. High priority packet data can only share high priority voice capacity, if idle circuit switched slots exist, and the moving boundary allocation policy is in use. Hence, priority 1 voice blocking probability under precedence ordering 2 can be expressed as:

$$B_{1,2} = B_{1,1} = B\ (a_1,S) \tag{87}$$

If sharing of the idle voice slot capacity by packet data is not permitted (fixed boundary slot allocation), no interaction among the different traffic types occurs, and the delay and blocking probability analyses are identical to the results outlined in the preceding section with:

$$\hat{\delta} = \frac{N}{N+[S/I]} \tag{88}$$

In the event idle voice slot capacity can be shared by the packet data, we postulate that high priority data can preempt low priority voice. Hence, only the high priority voice will influence the performance received by priority 1 packet data. The average priority 1 packet data delay therefore becomes:

$$D_{1,2} = \frac{t}{2} \frac{(2-\theta_1 t)}{(\sigma^* - \theta_1 t)} \tag{89}$$

where

$$\sigma^* = \frac{N + \left[\frac{S-\varepsilon^*}{I}\right]}{N + [S/I]} \tag{90}$$

and

$$\varepsilon^* = a_1 (1 - B_{1,1}) \tag{91}$$

is the average voice slot utilization due exclusively to the high priority voice. Since high priority data can preempt low priority voice, the priority 2 voice blocking probability is modified to account for the additional presence of priority 1 data traffic input to the channel with higher precedence:

$$B_{2,2} = B(q,S) = \frac{\frac{q^S}{S!}}{\sum_{i=0}^{S} \frac{q^i}{i!}} \tag{92}$$

where q is the cumulative offerred load due to all voice and the high priority data, i.e.:

$$q = a_1 + a_2 + \theta_1 t \tag{93}$$

The average delay analysis for the low priority packet data is formidable due to the non-preemptive relationship between low and high priority data traffic types. We therefore employ the

conservative approximation:

$$D_{2,2} \cong D_{2,1} = \frac{t}{2} \frac{(\Theta_1 t+2(1-\partial))}{(\partial-\Theta_1 t)\ (\partial-(\Theta_1+\Theta_2)t)} + t \tag{94}$$

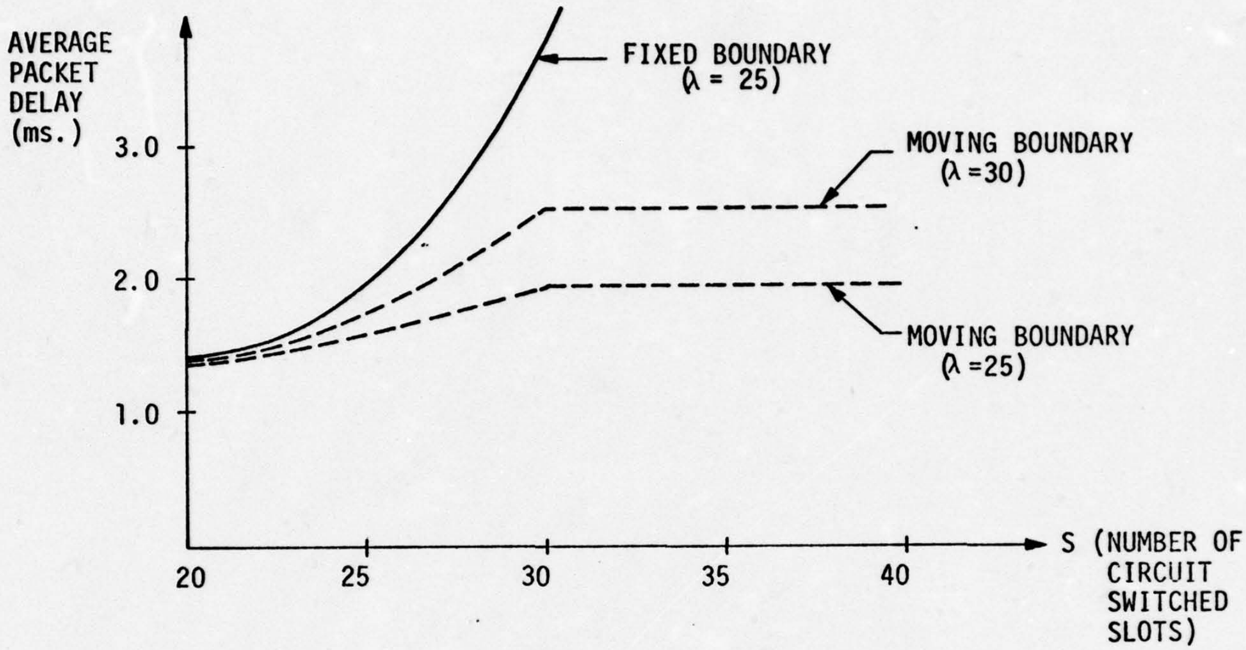
Under the moving boundary slot allocation policy for precedence ordering 2, Eq. (94) represents an upper bound on the low priority packet data delay; since the ability of high priority data to preempt low priority voice effectively "frees up" a greater portion of the available voice slot capacity, which can then be used by priority 2 packet data. Hence, approximate performance (blocking and delay) for the hybrid switching channel under precedence ordering 2 have been derived. If partial buffering of circuit-switched connection requests is provided (combined loss-delay system), non-preemptive operating rules for the circuit switched traffic could potentially exist, which would pose further analytic complexities. A final variant of the priority mechanism for the circuit switched voice places an upper bound on the number of voice slots $\hat{S} < S$ which can be seized by the low priority voice traffic. A straightforward analysis can be carried out for precedence ordering 1, however results for precedence ordering 2 are not readily obtained.

2.5 EXPERIMENTAL RESULTS

For convenience, voice traffic is assumed to be circuit-switched, and data traffic is assumed to be packet-switched; therefore, we may often use the terms voice and circuit, or data and packet, synonomously without loss of generality. Interesting aspects of the integrated switching channel's performance, that have emerged as a result of the experimentation, will be described qualitatively wherever appropriate; however, we must point out that the primary utility of the analytic models lies in our ability to quantify the channel's performance. All parameters used to obtain a particular graph are explicitly indicated.

2.5.1 Performance as a Function of Slot Allocation and Traffic Levels

Figure 13 shows the average packet delay as a function of number of slots allocated for circuit switched voice traffic (S) for the fixed and moveable boundary frame management policies, and parameters as indicated in the figure. In the fixed boundary case the average delay increases with S, whereas in the moveable boundary case the average delay levels off. The latter occurs when B $(\lambda h, S)$ becomes small for a constant λh and increasing values of S. A poor choice for the number of dedicated circuit switch slots S does not seriously impact the average packet delay under the moving boundary slot allocation policy. The average data packet delay at a heavier voice load (λ=30) is also shown for the moving boundary policy in Figure 13. As the voice traffic intensity increases, the threshold at which "leveling off" occurs also increases; furthermore, the number of voice slots (S) necessary to reach the threshold also increases at greater levels of voice traffic. The average delay encountered using the fixed boundary policy is independent of offered voice load.



FIXED PARAMETERS:

λ = 25 CALLS/MINUTE	C = 1.544 Mbps	P = 1000 BITS
h = 1 MINUTE	b = 10 ms.	Θ = 250 PACKETS/SEC.
VDR = 32 Kbps		

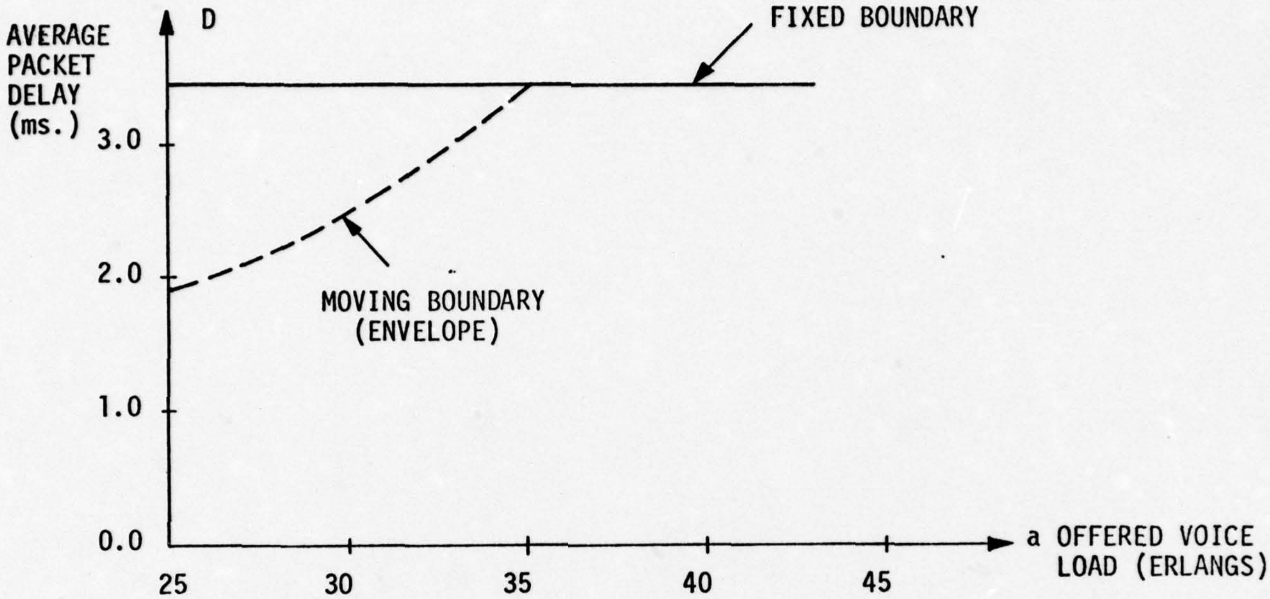
FIGURE 13: PERFORMANCE AS A FUNCTION OF CIRCUIT SWITCHED SLOT ALLOCATION

Figure 14 illustrates channel performance as a function of the offered voice load, the average packet data delay is identical for either slot allocation policy at high levels of voice traffic intensity. This arises because the incoming voice traffic will encounter high levels of blocking and consequently, individual slots are heavily utilized resulting in little surplus channel capacity available for data. As can be seen from Figure 14 the advantage of a moveable boundary frame management is eliminated at high voice traffic load; or alternatively when the link is engineered for a high voice blocking probability.

Finally, the average data packet delay as a function of packet arrival rate is shown in Figure 15 under a constant offered voice load and constant circuit switching slot allocation. For the representative values given, the voice slot utilization is sufficiently low so that the moving boundary slot allocation policy is consistently superior to the fixed boundary policy. Under worst case conditions (heavy voice load), the moving and fixed boundary slot allocation policies would yield similar performance. For a data arrival rate of $\Theta=250$, the fixed boundary policy only yields 25% higher delay than the moving boundary policy; for higher packet traffic intensity ($\Theta = 750$), the relative difference in performance increases to 30%.

2.5.2 Slot Assignment Techniques

In this section, we investigate the impact of changing the slot assignment on channel performance as a function of traffic. Throughout the remainder of this chapter, we will effect a distinction in nomenclature between slot allocation and slot assignment. The former refers to the manner in which channel capacity is shared and controlled on a real-time basis (fixed or moving boundary); the latter will describe the manner by which the actual number of slots per frame are assigned to a particular traffic category (i.e., the boundary location.)



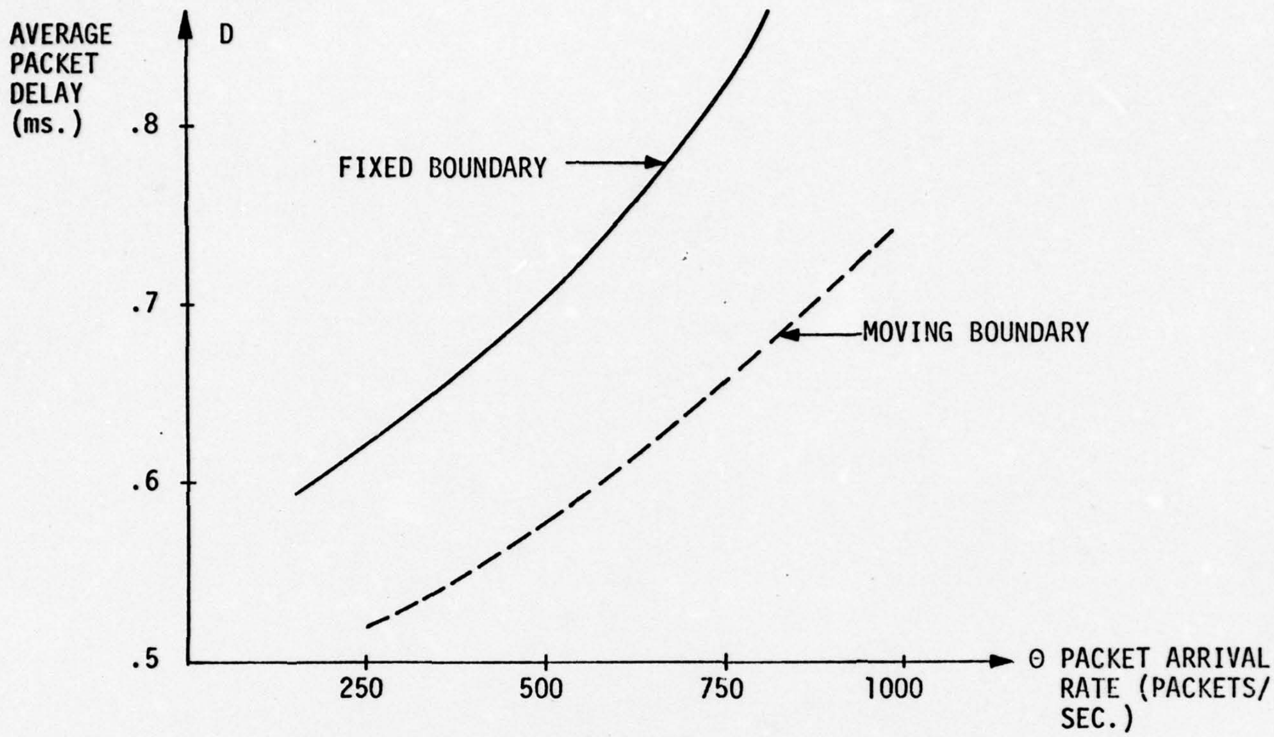
FIXED PARAMETERS:

S = 30
h = 1 MINUTE
VDR = 32 Kbps

C = 1.544 Mbps
b = 10 ms.

Θ = 250 PACKETS/SEC.
P = 1000 BITS

FIGURE 14: PERFORMANCE AS A FUNCTION OF OFFERED VOICE LOAD



FIXED PARAMETERS:

C = 3.088 Mbps	P = 1000 BITS	λ = 30 CALLS/MINUTE
b = 10 ms.	h = 1 MINUTE	S = 40
		VDR = 32 Kbps

NOTE: THE CIRCUIT SWITCH BLOCKING PROBABILITY REMAINS FIXED AT $B \cong .0144$

FIGURE 15: PERFORMANCE AS A FUNCTION OF INPUT PACKET DATA TRAFFIC INTENSITY

The fraction of channel capacity which is assigned to handle each traffic type (circuit (p_1) and packet (p_2)) can be determined in several ways. In general, given the desired performance constraints (required average delay, D', and required blocking probability, B'), offered traffic (a erlangs, Θ packets/sec.) and a channel of fixed total capacity C, a compound performance objective function must be minimized by the assignment of the available capacity (p_1,p_2).
Specifically; the problem is to determine p_1 and p_2 which

$$\text{Minimize}\left[\alpha(B(a,p_1)-B')^2 + \beta(D(\Theta,p_2)-D')^2\right] \qquad (95)$$

subject to $p_1 + p_2 \leq 1$, where α and β are weighting functions, and, $B(a,p_1)$ and $D(\Theta, p_2)$ are the traffic blocking and delay respectively, which result from the capacity assignment (p_1,p_2). It is not our purpose in the current investigation to find the preceding "optimal" assignment. Rather, we will examine several potential slot assignment schemes including: Proportional capacity assignment (based on the offered traffic intensity), slot assignment under a fixed loss constraint (β=0), and slot assignment under a fixed delay constraint (α=0). A characterization of three slot management schemes is provided in the Appendix C.

2.5.2.1 Proportional Slot Assignment

A simple slot assignment scheme, which effectively partitions the channel capacity in proportion to the traffic input to the switching system is now examined. Specifically, channel capacity is assigned in proportion to the total amount of offered traffic belonging to a particular class. Hence, the fraction of capacity dedicated to circuit switching (voice) is given as:

$$p_1 = \frac{a\ VDR \cdot b}{a\ VDR \cdot b + \Theta P} \qquad (96)$$

and the fraction dedicated to packet switching (data) is $p_2=1-p_1$.

We do not imply that this scheme is to be preferred over others; in fact, as will be shown, under different traffic mixes, the above fractions do not yield desirable levels of performance.

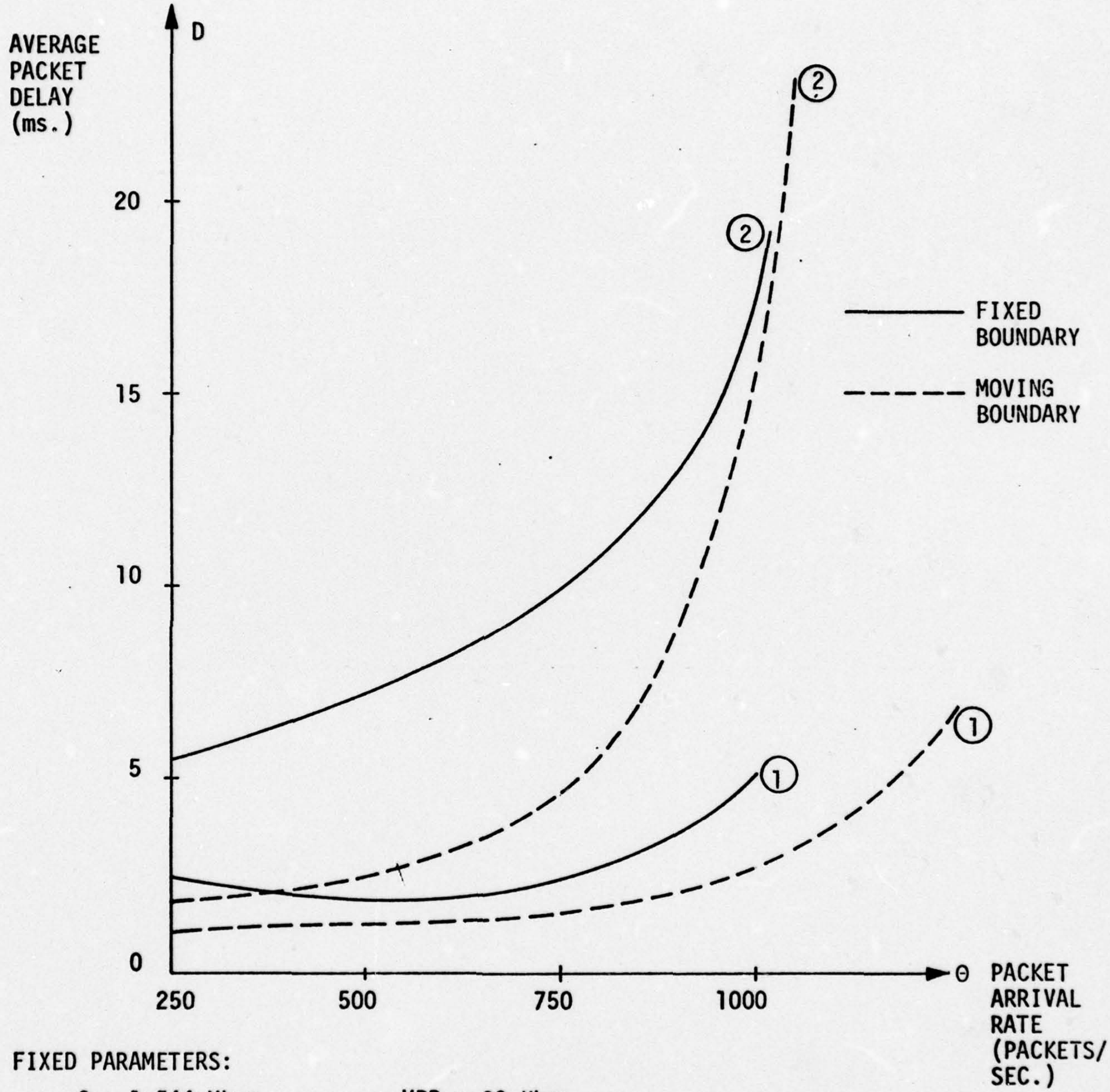
Ideally under Proportional Slot Assignment scheme (PSA), the portion of the channel capacity relegated to each traffic type would be p_1C (p_2C). In general, however, due to the frame structure and because the packet length and voice slot size are not integral multiples of one another, the ideal capacity partition cannot be achieved. For example, if the PSA dictates that 6720 bits per frame be allocated to data, with 1000 bit packets, only a maximum of 6 data slots may be used (assuming no packet fragmentation is allowed). This presence of "slack" capacity (i.e., 720 bits) suggests two simple approaches of controlled sharing between both competing traffic classes; giving preference to data traffic or to voice traffic. The latter will be assumed in the numerical results.

The average delay encountered by data packets input to the integrated switching channel under PSA is depicted in Figure 16, as a function of the packet arrival rate. It is noted that the values of S and N are changing with Θ according to Eq.(96). The qualitative behavior shown in Figure 16 is expected, within the variations resulting from the assignment of the "slack" capacity.

The grade of service provided to circuit switching voice traffic as a function of the packet arrival rate and erlang load is shown in Figure 17. Under increasing data load, blocking increases due to the reduced channel capacity for voice (Figure 17). Under increasing voice load, even though the channel allocation increases, the slots become more heavily utilized and blocking increases.

2.5.2.2 Constraint-Based Slot Assignment

The performance rendered to traffic classes input to the integrated switching channel using a second class of slot assignment

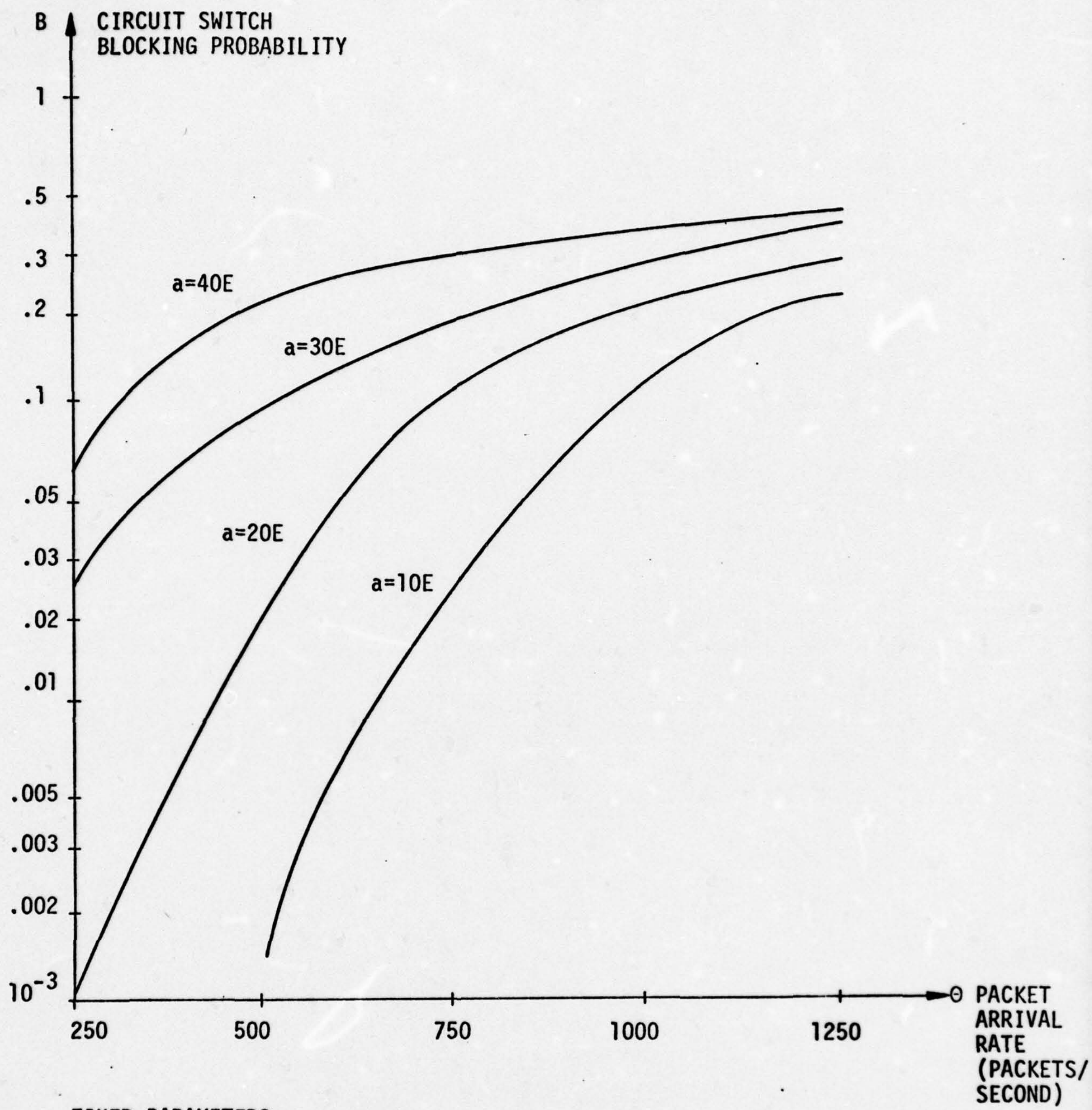


FIXED PARAMETERS:

C = 1.544 Mbps VDR = 32 Kbps
b = 10 msec. P = 1000 BITS h = 1 MINUTE

CURVES: 1 VOICE LOAD = 10 ERLANGS
2 VOICE LOAD = 20 ERLANGS

FIGURE 16: AVERAGE PACKET DELAY UNDER THE PROPORTIONAL CAPACITY SLOT ASSIGNMENT POLICY VS. PACKET ARRIVAL RATE



FIXED PARAMETERS:

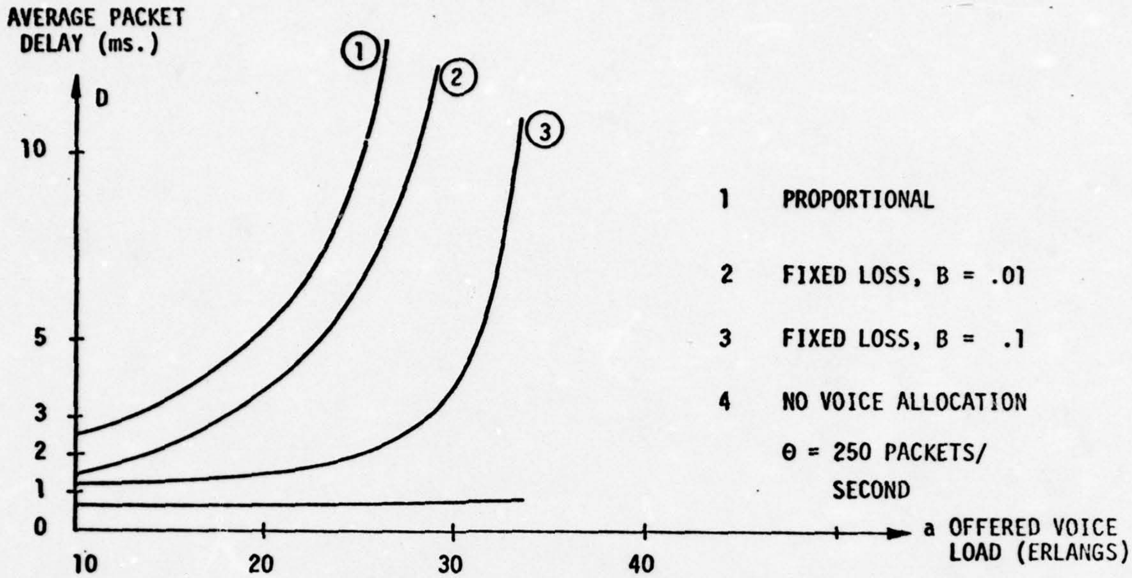
C = 1.544 Mbps VDR = 32 Kbps h = 1 MINUTE
b = 10 ms. P = 1000 BITS

FIGURE 17: CIRCUIT SWITCH BLOCKING PROBABILITY UNDER THE PROPORTIONAL SLOT ASSIGNMENT POLICY VS. PACKET ARRIVAL RATE

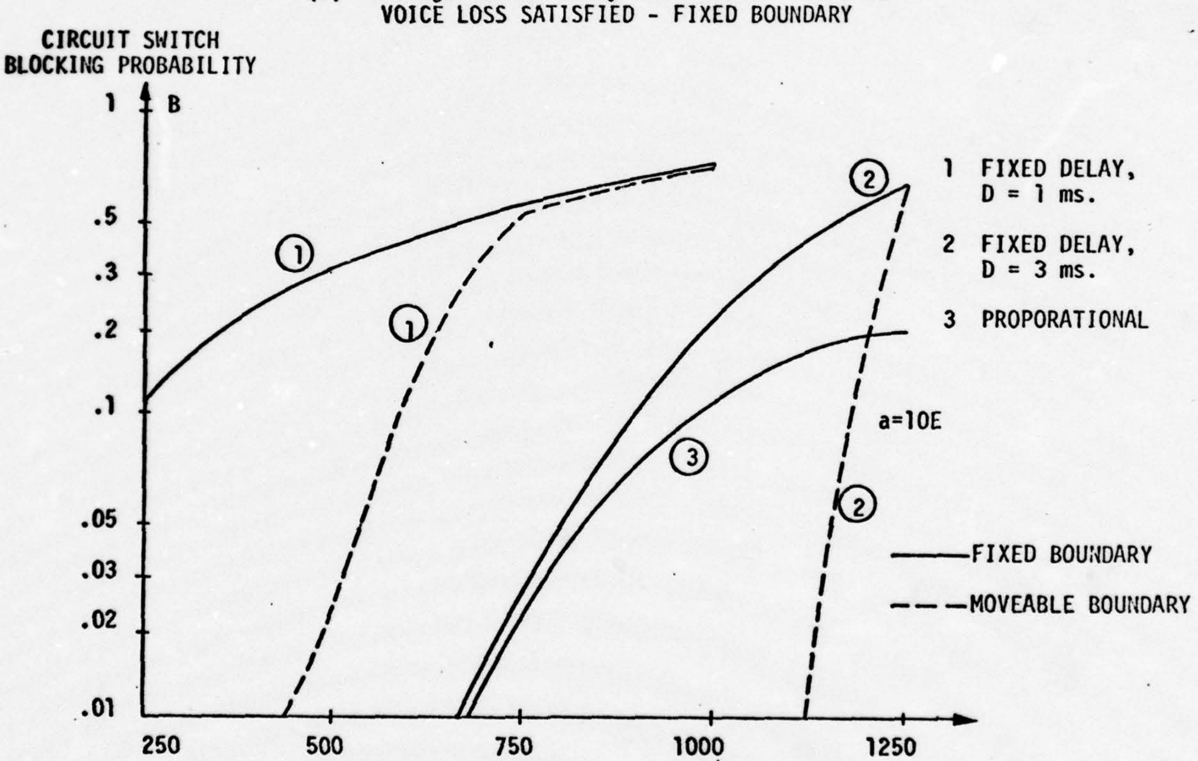
techniques is now discussed. The motivation underlying these so-called "constraint-based" techniques is to give preferential treatment to one class of traffic. For example, a delay constrained slot assignment technique provides an adequate number of data slots to handle incoming packet traffic at a fixed level of performance (α = 0, in Equation (95)); similarly, a blocking constrained slot allocation technique provides sufficient circuit-switching capacity (voice slots) to yield a uniform grade of service for incoming speech (β = 0, in Equation (95)). Clearly, under certain traffic mixes and fixed channel capacity, both performance constraints may be simultaneously satisfied and hence, increased channel capacity may often be required to obtain acceptable service. In this section, we investigate the impact a constraint-based slot allocation policy (for either circuit voice or packet data) exerts on the performance of the other traffic type. For all examples illustrated, the constrained performance parameter (e.g., a blocking probability equal to .02) could often not be exactly achieved. Therefore, the slot allocation which most closely satisfies the desired requirement was used, with the constraint acting as an upper bound on performance.

Figure 18, shows the performance obtained under the different slot assignment techniques. Figure 18a contrasts the average packet delay encountered under the proportional slot assignment technique (PSA) (Curve 1) and the blocking-constrained slot assignment technique (Curves 2 and 3). Figure 18b depicts the blocking probability for circuit switched voice under PSA and the delay-constrained slot assignment technique for the fixed and moveable boundary frame management.

The PSA is shown for comparison, and its performance relative to the other assignment schemes depends on the total available capacity C. It should be emphasized that the slot allocation policies which have been proposed may often only be of interest from an analytic standpoint. In reality, dual sets of performance constraints must be satisfied, and this will usually require upgrading the channel capacity as well as perturbations to the slot assignment.



(a) Average Packet Delay vs. Offered Voice Load
VOICE LOSS SATISFIED - FIXED BOUNDARY



(b) Blocking Probability vs. Data Traffic Intensity
DATA DELAY SATISFIED

FIGURE 18: COMPARISON BETWEEN SLOT ASSIGNMENT POLICIES

2.5.3 Additional System Parameters

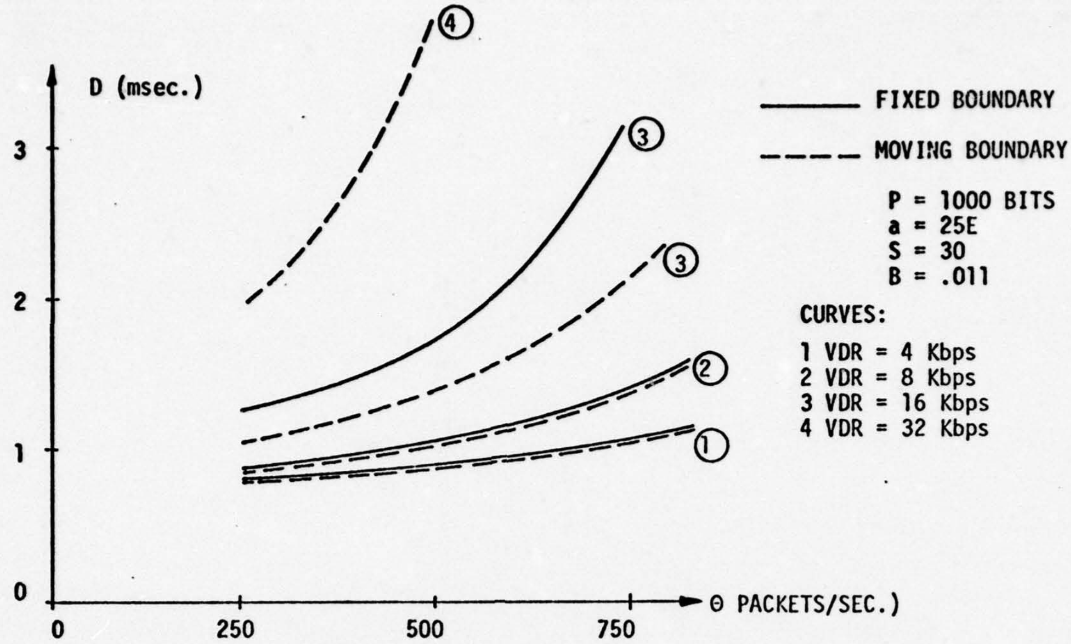
The performance provided to packet and circuit switching subscribers by the integrated channel depends on several additional parameters apart from the slot allocation/assignment policies and input traffic intensity. The parameters on which we now focus attention include:

- Voice digitization rate.
- Speech activity factor.
- Packet length.

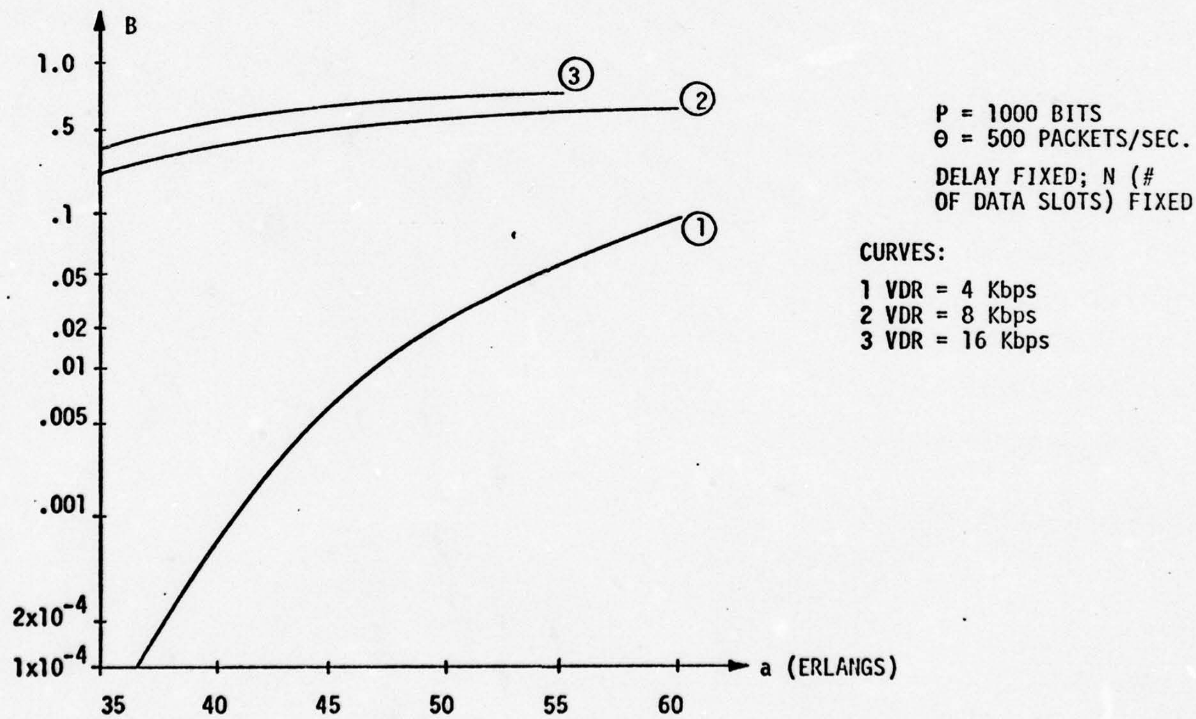
In particular, the impact of each parameter on channel performance is quantitiatively examined. All three parameters are often assumed to be"givens", i.e., they are characteristic of the subscribers, based on their existing equipment (e.g.,voice digitization and speech interpolation hardware, packet segmentation software, etc.); their intrinsic presence must be dealt with in any system design. To the extent that these parameters are also under the control of the system architect, they represent additional dimensions to the design problem, since an appropriate choice of VDR and P or the use of speech interpolation can affect significant cost-performance benefits. Of course, as part of the total network design problem, any advantages obtained from a particular choice of VDR, P or the use of speech interpolation, such as increased channel efficiency and/or reduction in bandwidth requirements, must be weighed against the cost of the devices required to achieve these features. For example, at lower values of VDR, the hardware needed to generate the digitized speech may become appreciably more expensive. Similarly, devices which monitor speech activity (in order to implement speech interpolation) are also of significant cost.

In considering VDR, we assume that all voice (which is circuit switched over the integrated channel) is digitized at the same rate. In actuality, there exist several classes of voice (digitized using different strategies), which can potentially be switched over the same channel. The channel performance is shown in Figure 19 for several different digitization rates. In Figure 19, we have plotted the average delay for data traffic under different voice digitization strategies as a function of the packet arrival rate. Since, the number of circuit-switched slots is fixed, as the voice digitization rate increases, the size of an individual voice slot will increase. Consequently, the portion of total frame capacity that is allocated to voice becomes larger. Note that the blocking (B=.011 in Figure 19) remains constant, since no additional slots are added, but each slot is increased in length. Hence, for fixed S, as VDR increases, the number of data slots will decrease, and therefore, the average packet delay must increase. For example, under the moving boundary slot allocation policy with $\Theta = 500$, a=25E(Erlangs), the average packet delay is .94 msec. when VDR=4 kbps, but increases to 3.8 ms. when VDR=32 Kbps.

A second interesting aspect of the average delay (Figure 19) is that at relatively low voice digitization rates, the fixed and moving boundary slot allocation policies yield similar performance. In the prescence of larger VDR, a difference between fixed and moving boundary policies begins to emerge. The reason is due exclusively to the packet size and the circuit switch slot loading. Although the blocking may be relatively low on the average, at low values of VDR, the individual slots are relatively small and cannot be combined to equal the required size of a data slot (packet length). Therefore, the moving boundary allocation policy cannot make use of the spare capacity, and the identical level of performance with the fixed boundary is obtained. As shown in Figure 19, the average packet delay is identical under either fixed or moving boundary policies for VDR=4 Kbps and 8 Kbps; when VDR increases to 16 Kbps, the average



(a) Average Packet Delay as a Function of Voice Digitization Rate



(b) Circuit Blocking Probability as a Function of Voice Digitization Rate

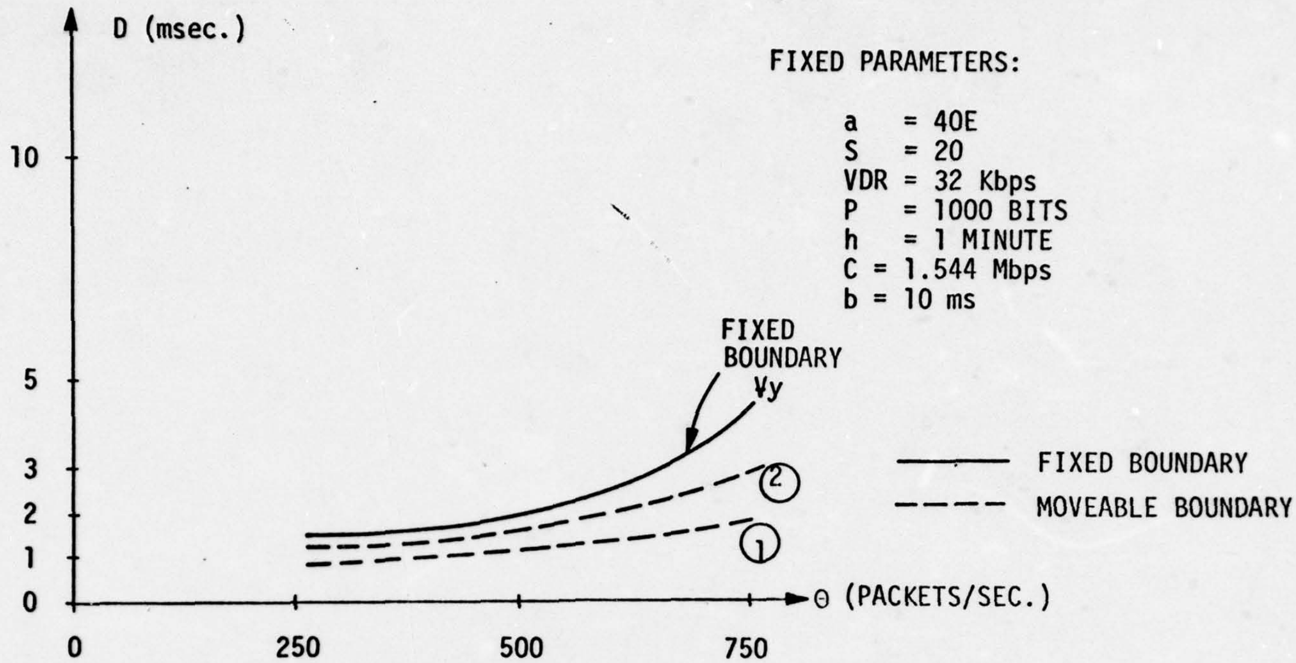
FIGURE 19: PERFORMANCE DEPENDENCY ON VOICE DIGITIZATION TECHNIQUE

packet delay is greater under the fixed boundary policy by .2ms. at Θ=250 and over 1 ms. at Θ=750. The choice (if available as an option to the channel designer) of packet length should be made carefully in conjunction with the voice digitization rate, since the system performance depends on their relationship.

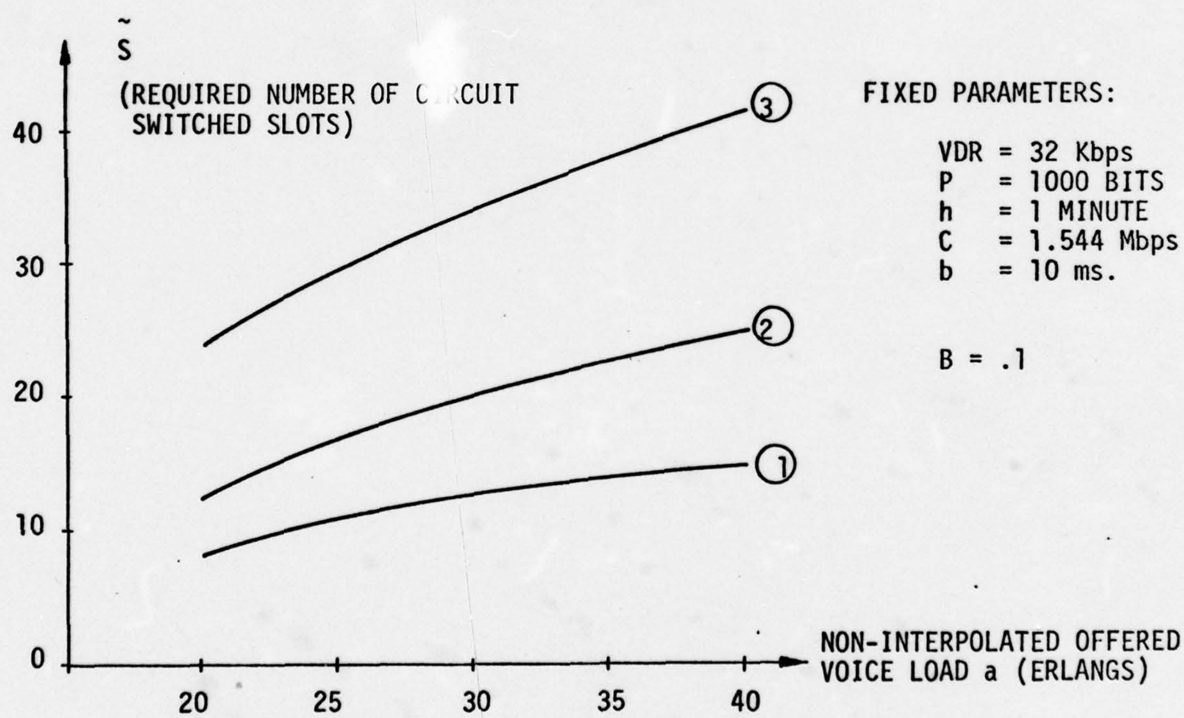
Figure 19b illustrates the circuit switched blocking probability for input voice calls as a function of VDR with the number of data slots fixed. Once again, because the amount of frame capacity which can be potentially used by voice is fixed by N, a low VDR implies <u>many</u> small voice slots with a correspondingly low blocking probability (Curve 1), whereas a high VDR would result in <u>few</u> large voice slots thereby yielding a poorer grade of service (Curve 2 and 3). At 40E (40 Erlangs) offered voice load, the blocking probability is less than .001 when VDR=4 Kbps, but exceeds .5 (a factor of 500 increase) when the digitization rate is increased by a factor of four to VDR=16 Kbps.

The impact on channel performance of speech interpolation is shown in Figure 20 as a function of the speech activity factor y (percentage of time a talker is active). Typical values of y for a single talker are y=.25 (normal conversation), y=.5 (reading). Obviously, any improvement to data traffic's performance due to a speech interpolation capability can only be gained if the moveable boundary slot allocation policy is employed, since only then will idle circuit switched slots in the frame be made available for sharing by data traffic. In Figure 20a, as the speech activity factor decreases, the number of temporarily idle voice slots in a frame increases, providing more potential capacity for data packet transmission. Thus, at y=.5, Θ =750, the average packet delay D=2.8 ms., but is reduced to 1.5 ms. when y decreases to .25.

Speech interpolation also offers obvious advantages to the circuit switched voice traffic performance, since idle slots can be used to carry additional "fresh" incoming calls. Figure 20b demonstrates that for a fixed grade of service (B=.1), the



(a) Average Delay vs. Data Traffic Intensity



(b) Channel Capacity Allocation Required to Attain a Fixed Level of Loss
CURVES: 1 SPEECH ACTIVITY FACTOR = y = 25%
2 SPEECH ACTIVITY FACTOR = y = 50%
3 SPEECH ACTIVITY FACTOR = y = 100%

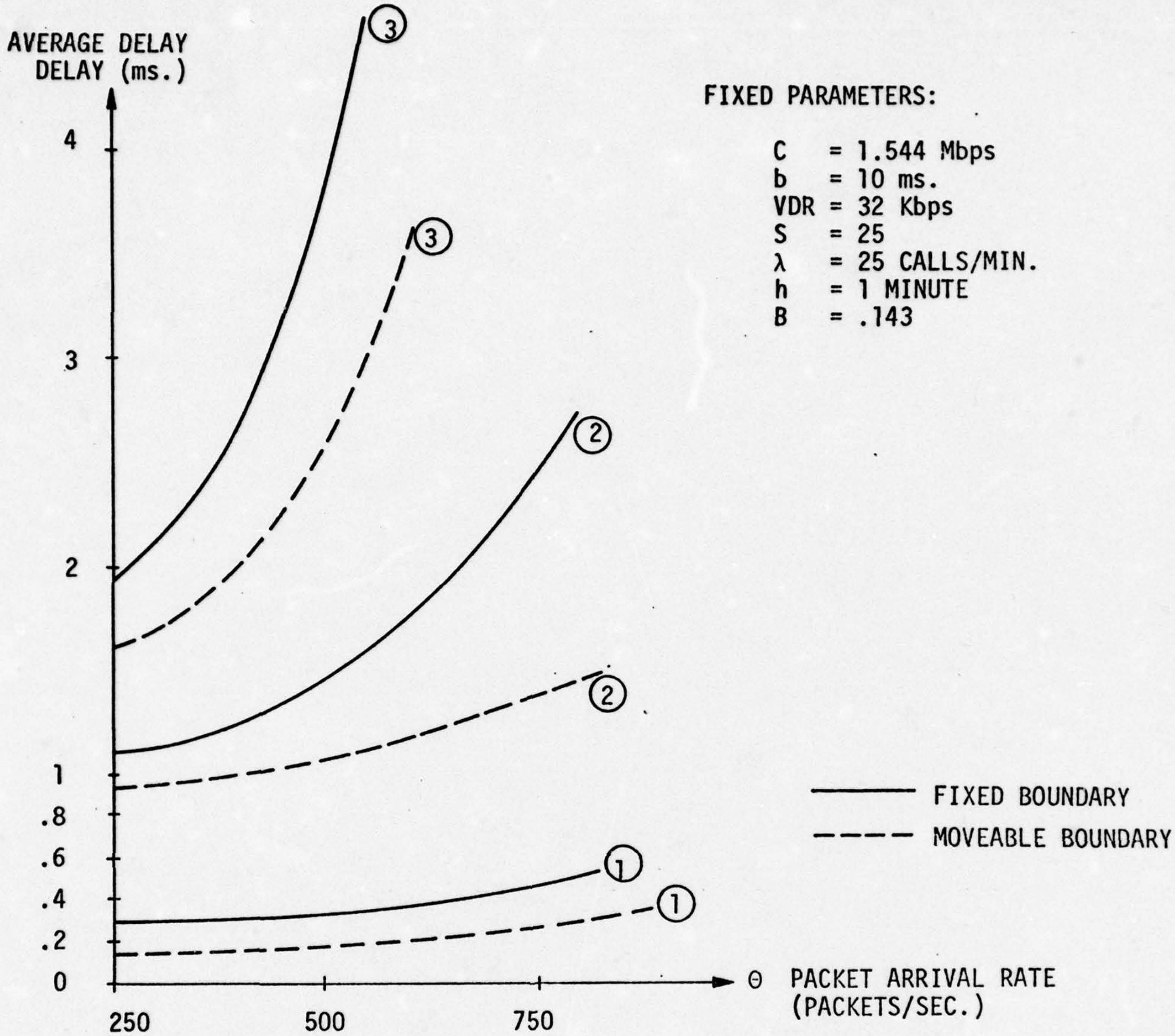
FIGURE 20: THE IMPACT OF SPEECH INTERPOLATION

total number of circuit switched slots S (capacity requirement) needed to sustain an incoming offered load is lower as the speech activity factor decreases; at a=40E, B=.1, the number of slots required for y=.5, (S=25), is somewhat less than double the number required for y=.25, (S=14). It should also be emphasized that the use of speech interpolation depends in part on the digitization technique used to encode the voice; several of the low data rate technologies (e.g.,vocoders) require periodic transmission of the spectral content of the speech waveforms, other techniques such as delta modulation, inherently do not occupy channel bandwidth if significant waveform variations occur. [OCCHIOGROSSO, 1976]

The choice of packet length is an important design variable in any store-and-forward communication switching system. When transmitted over an integrated switching channel, the decision is further complicated, since the voice and data slot size are in a sense interrelated with respect to performance. Figure 21 depicts the average delay for incoming data packets as a function of packet length. Longer packets suffer a greater delay than shorter packets, as is to be expected. Under the moving boundary at Θ=500, D=2.6 ms. when P=1000 bits, but is reduced to less than .2 ms. for P=250 bits. The choice of packet length does not in any way influence the circuit switching grade of service.

One criterion for determining packet size is to minimize the number of slack unused bits in a frame (no fragmentation of packets is permitted over several frames), while at the same time not allowing the ratio of packet header to information content to become excessively large. The situation described is similar in flavor to the classic "packing" problem. Therefore, in selecting a suitable packet length for transmission over the integrated channel, cognizance should be taken of the following issues:

- Size of incoming data messages.



CURVES: 1 PACKET LENGTH = P = 250 BITS
2 PACKET LENGTH = P = 500 BITS
3 PACKET LENGTH = P = 1000 BITS

FIGURE 21: DATA PERFORMANCE AS A FUNCTION OF PACKET SIZE

- Excessive queueing delay due to long packets.

- High degree of overhead due to small packets.

- Inefficient channel operation under the moving boundary slot allocation policy, if the packet size is significantly greater than the voice slot size.

- Frame duration and the possibility of packet fragmentation.

2.5.2.3 Behavior of the Signaling-Regular Packet Traffic Mix and Approximation by Single Class Delay Formulas

In this section, we examine the behavior of the average packet delays in the context of mixing the circuit signaling traffic and the regular packet traffic. We assume that the signaling packet size is on the order of 100-200 bits, and that the signaling packets constitutes (class 1) a small fraction of the total packet traffic (say, less than 20% of the total traffic flow). Class 2 traffic refers to regular packets. Other parameters used in the numerical results are C=1.5 Mbps, b=10 m sec, VDR = 50 Kbps, λ'=10 calls/sec. h=1 sec, and S=20. The moveable boundary frame management strategy is used.

Figure 22 plots the average packet delay T_2 as a function of the average packet arrival rate Θ_2 for the class 2 packets, with Θ_1 as a parameter (based on Eq.(55)) for the Head of the Line, non-preemptive Priority case. It can be seen that;

1. if the channel utilization is not too high (say, < 70%), or if Θ_1 is small (say, $\leq$ 200 packets/sec), then a reasonable estimate for T_2 can be obtained by disregarding the class 1 flow (i.e.,setting Θ_1 to 0) and reducing the problem to the single packet class delay problem;

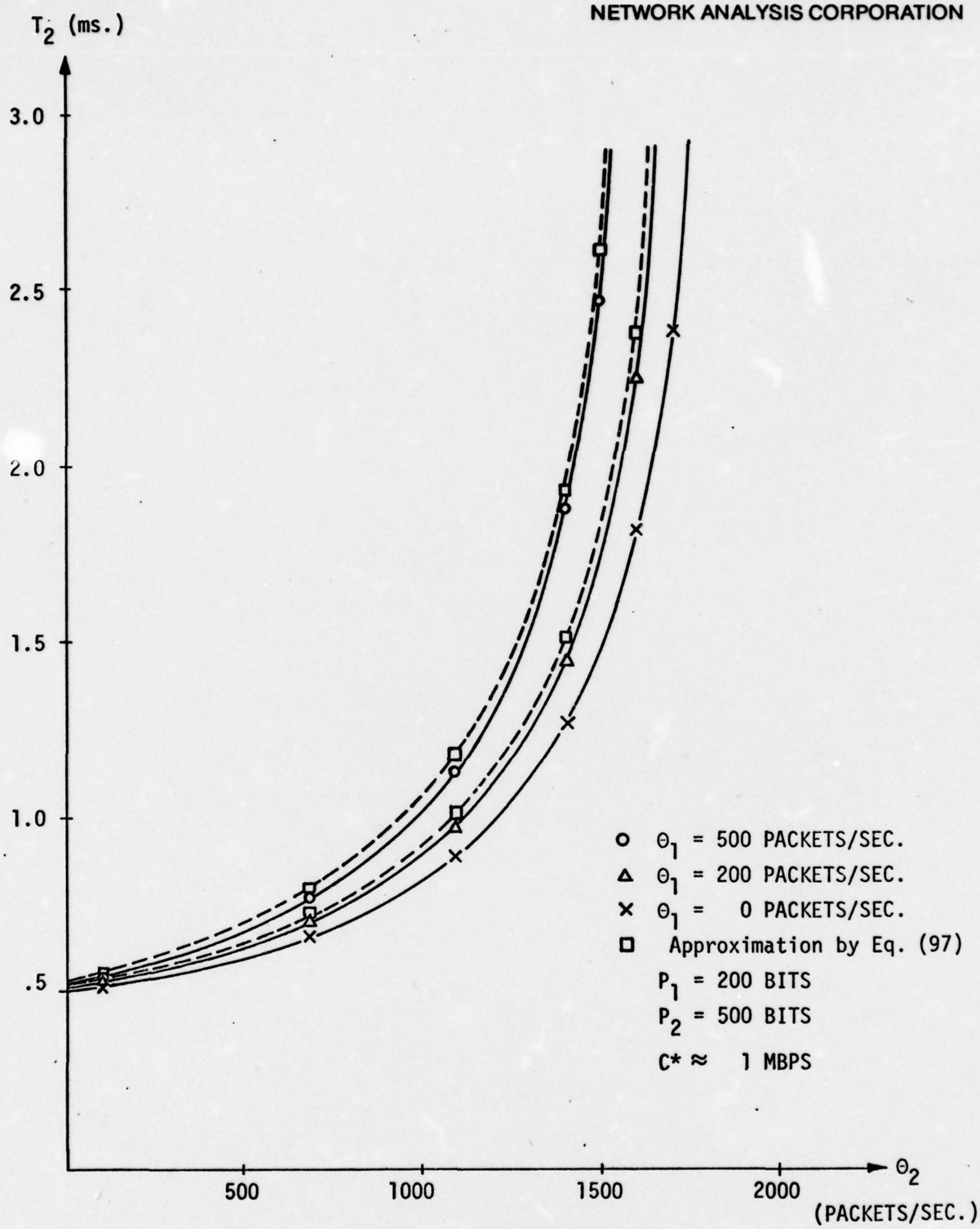


FIGURE 22: CLASS 2 PACKET DELAY VS. ARRIVAL RATE PRIORITY, HOL, NON-PREEMPTIVE

2. T_2 is usually much larger than T_1.

The average delay estimation for the class 2 packets by ignoring class 1 flows can be justified only if the class 1 packet traffic is very small. If we increase the class 2 packet flow to include also the class 1 flow (but keep the class 2 packet size unchanged), and decrease the class 1 packet flow to 0, we obtain another approximation for the average class 2 packet delay using only the single packet class queueing formula.
This approximation can be obtained by subtitution into Eqs. (57) and (58), to yield:

$$T_2^{\,1} = \frac{P_2[2C^* - \sigma(f_1+f_2)]}{2C^*(C-f_1-f_2)}. \qquad (97)$$

This approximation can analytically shown to be "good" if $C^* \gg f_1$. The approximation due to Eq. (97) is also plotted in Figure 22, where a close fit to using two packet types can be observed.

Assuming the proportion of signaling traffic is small compared to the regular packet traffic; by assigning signaling packets higher priority over regular packets, we can essentially ignore the signaling traffic (and modify the regular packet traffic accordingly) in the network design phase, and need only to evaluate its performance in the network analysis phase.

Figure 23 plots the average packet delay T_2 as a function of the average packet arrival rate Θ_2 for the class 2 packets, with Θ_1 as a parameter, for the non priority case (Eq. (71)). Same observations as in the priority case can be made; if the channel utilization is not too high (say, $\leq 70\%$) or if class 1 packet arrival rate Θ_1 is small (say, $\Theta_1 \leq 200$ packets/sec), then a good estimate for T_2 can be obtained by disregarding the class 1 flow (i.e., setting Θ_1 to 0).

Figure 24 plots the average packet delay T_1, as a function of the average packet arrival rate Θ_1 for the class 1 packets, with Θ_2 as a parameter, for the non priority case. As can

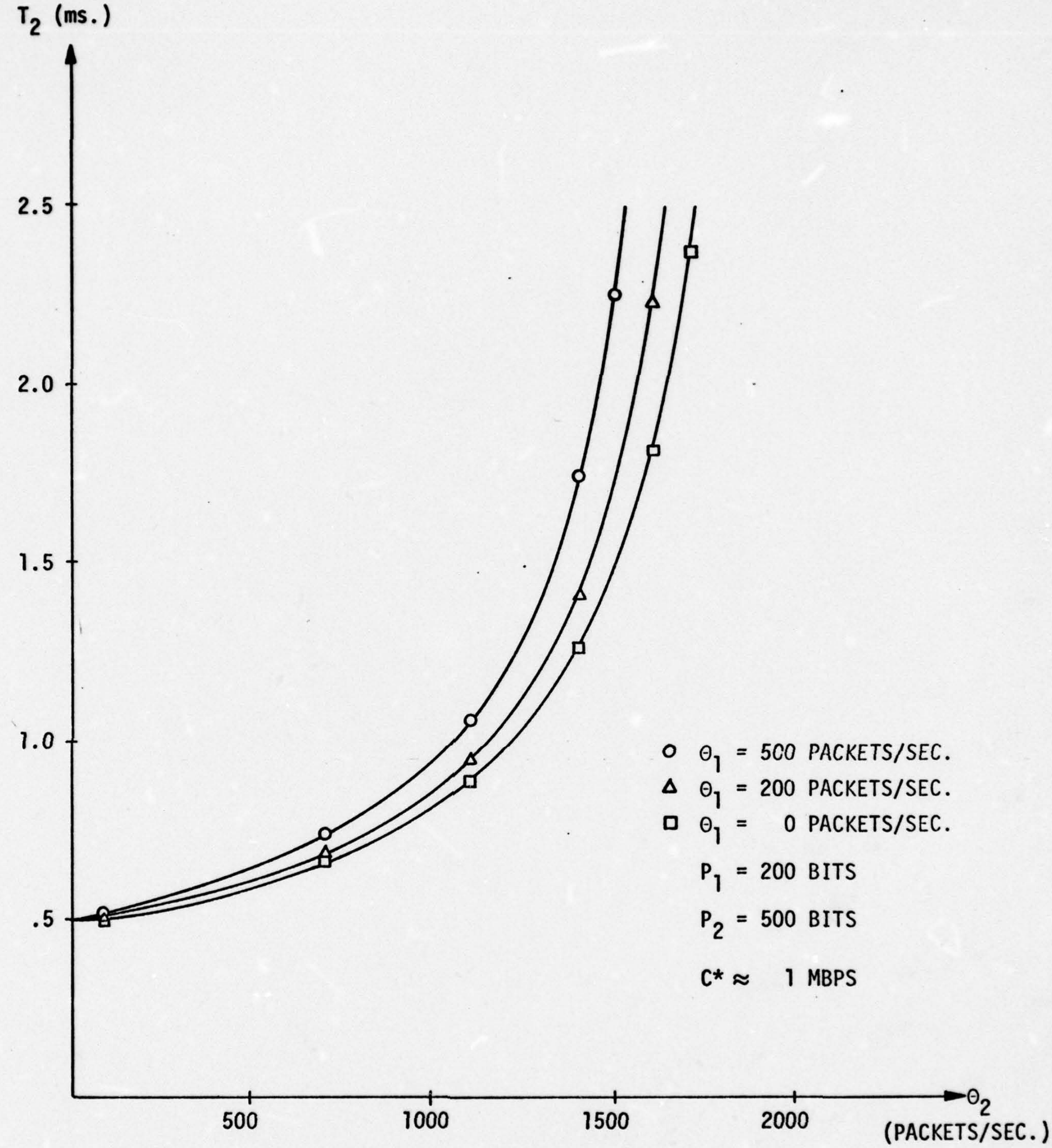


FIGURE 23: CLASS 2 PACKET DELAY VS. ARRIVAL RATE NON-PRIORITY DISCIPLINE

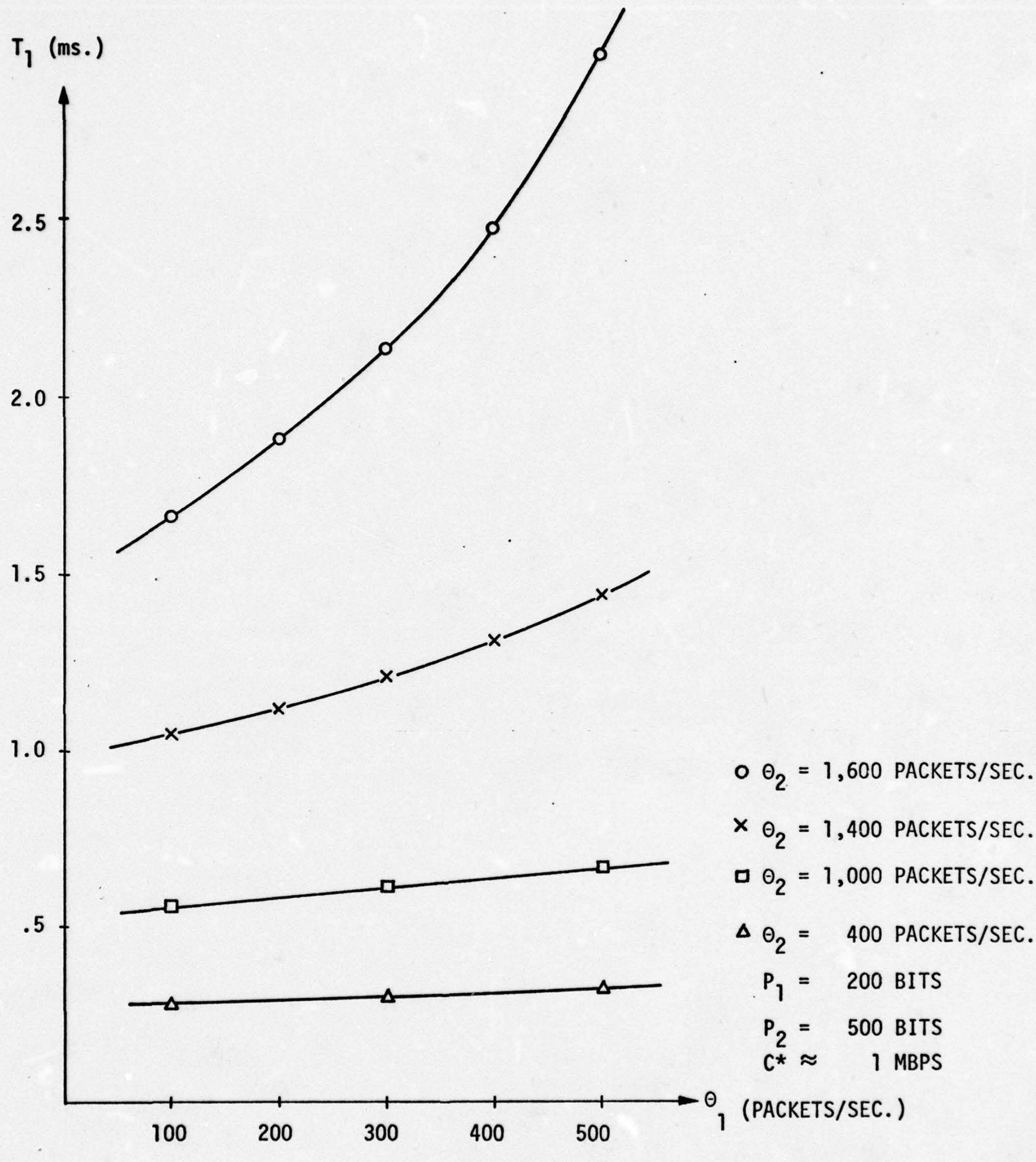


FIGURE 24: CLASS 1 PACKET DELAY VS. ARRIVAL RATE NON-PRIORITY DISCIPLINE

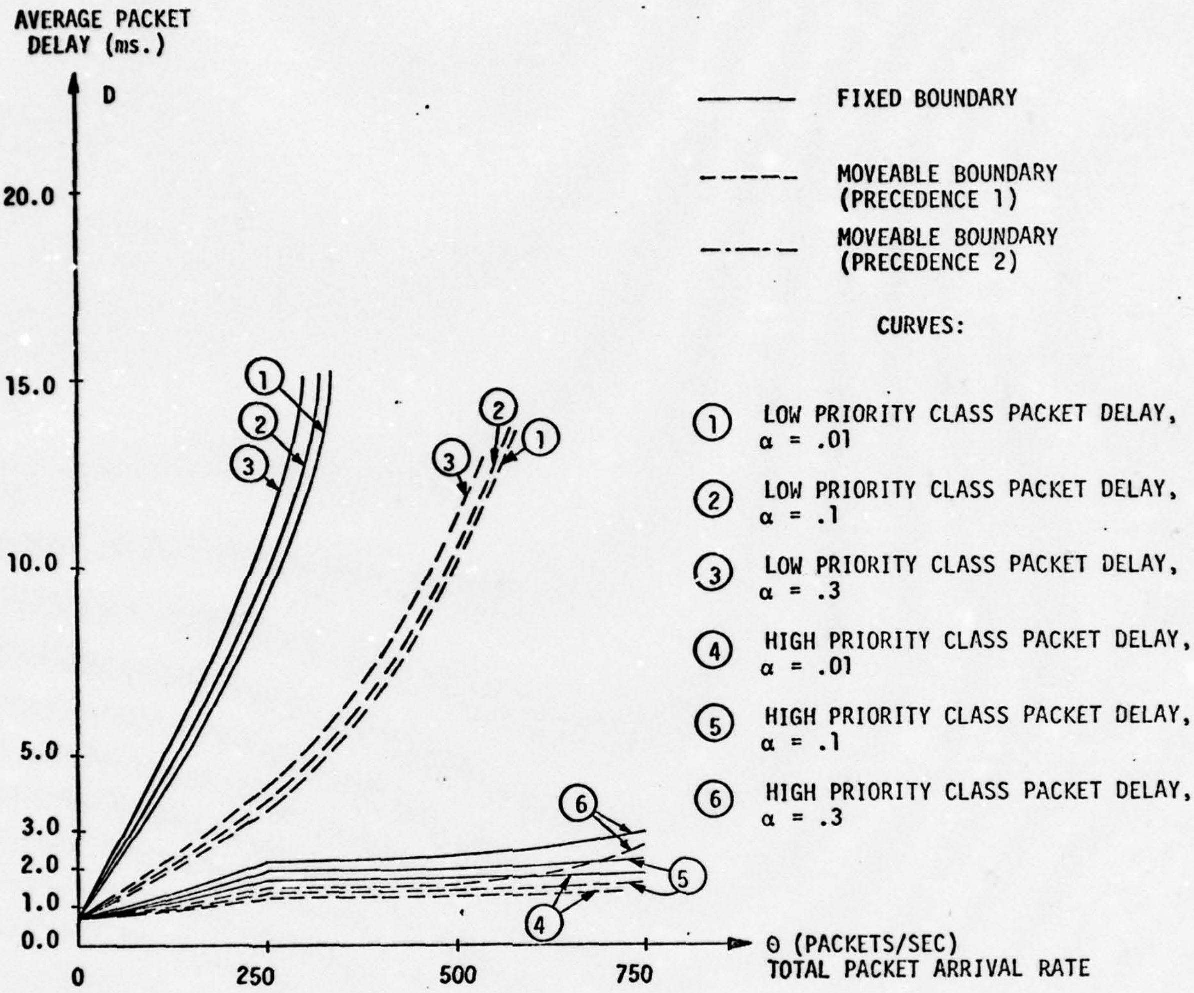
be seen, for small Θ_2, T_1 increases with Θ_1 quite rapidly. Hence, contrary to the priority case, T_1 can no longer be bounded within a small range. However, notice that both class 1 and class 2 packets have the same queueing delay, and the class 2 (i.e., regular) packets are longer than the class 1 (signaling) packets. Consequently, the average class 1 packet delay is always bounded from above by the average class 2 packet delay.

From the foregoing, we conclude that in network design, assuming that the proportion of the signaling traffic is small compared to the regular packet traffic; then if the average end-to-end delay constraints for the signaling traffic and the regular packet traffic are comparable, we can essentially ignore the signaling traffic (and modify the regular packet traffic accordingly) in the network design phase, and need only to evaluate its performance in the network analysis phase.

2.5.2.4 Voice and Data Priority Traffic Classes

Priority classes are expected to exist in the integrated switching environment for both the circuit-switched and packet-switched transactions. Two potential precedence orderings among the various priority classes: precedence Strategy 1 according to traffic types (voice or data) and precedence Strategy 2 according to priority class (high or low) were analysed in Section 2.4. We now investigate the quality of service received by traffic transmitted under an integrated switching discipline in a priority-based environment. Several aspects of the quantitative results to be presented are both a consequence of the analytic models and their implicit assumptions in addition to the actual system operation.

Figure 25 depicts the average packet delay with priority traffic as a function of the packet arrival rate Θ and the number of slots S which are dedicated to circuit switching of voice. In each case, both precedence strategies are examined and the percentage of high priority traffic α (for both voice and data) is varied.



Average Data Packet Delay as a Function of the Percentage of High Priority Traffic and Precedence Strategy

FIXED PARAMETERS:

C = 1.544 Mbps	λ = 30 CALLS/MINUTE (TOTAL)
b = 10 ms.	h = 1 MINUTE
S = 30	VDR = 32 Kbps
	P = 1000 BITS

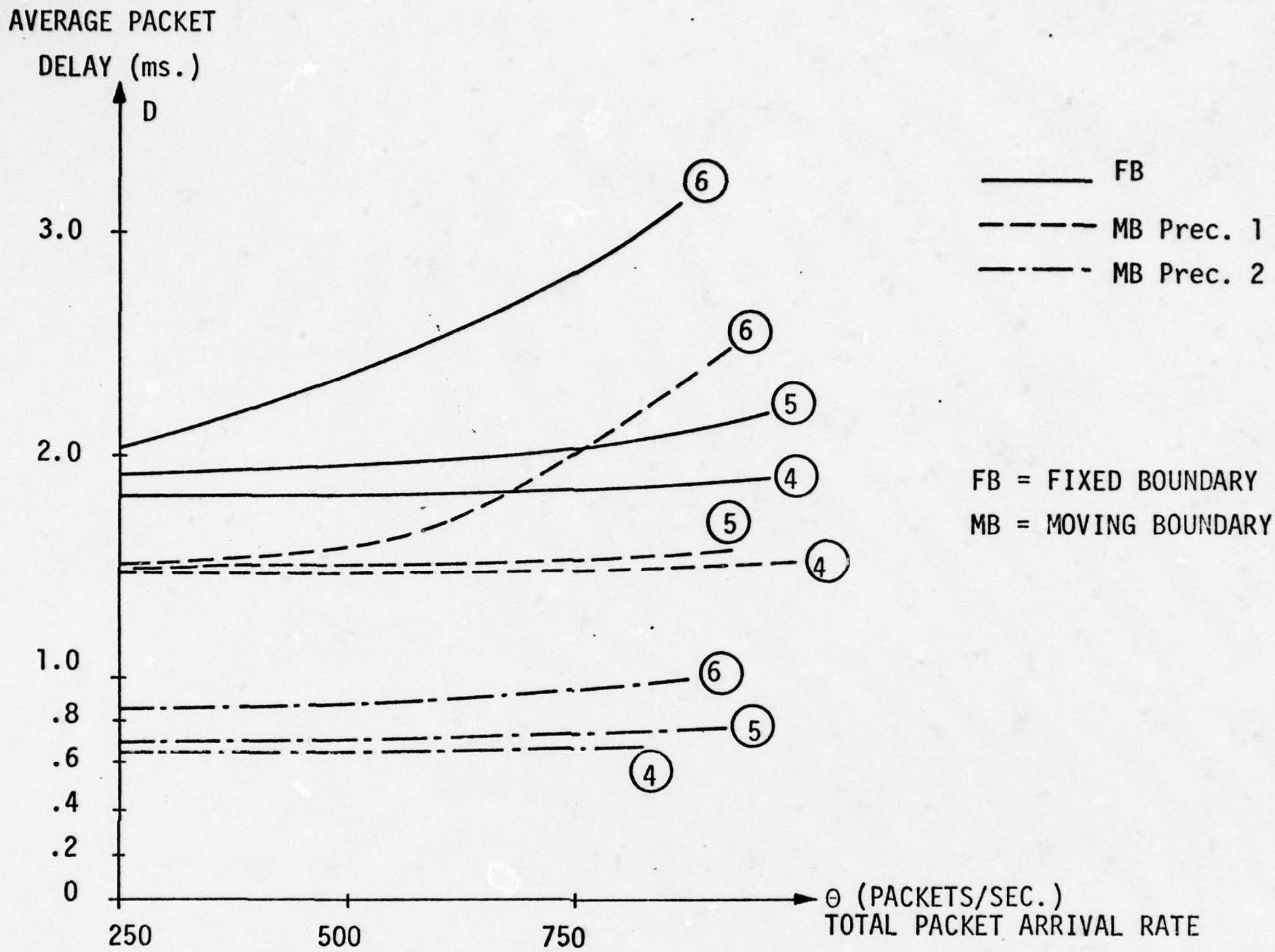
FIGURE 25: PACKET SWITCHING PERFORMANCE IN A PRIORITY BASED ENVIRONMENT FOR THE INTEGRATED CHANNEL

In Figure 25, the average delay for low priority packets is reduced substantially under the moving boundary slot allocation policy (a decrease of 6.25 ms. for $\alpha = .3$, $\Theta = 250$); however, the high priority packets receive only a marginal improvement using the moving boundary (approximately .4 ms. for $\alpha = .3$, $\Theta = 250$). An expanded diagram of the system performance for high priority traffic is supplied in Figure 26. Due to the relatively small percentage of high priority traffic (less than 30%) the low and high priority packet delay curves do not vary appreciably as a function of Θ.

Based on the earlier derived models, the low priority packet delay is identical under both precedence strategies. The average delay encountered by high priority packets is, however, reduced under precedence ordering 2, which favors high priority data packets over low priority voice calls. In fact, if the moving boundary slot allocation policy is employed, at $\Theta = 750$, $\alpha = .3$, precedence ordering 1's packet delay is 1 ms. higher than the corresponding performance associated with precedence ordering 2.

Figure 27 shows the data traffic delay for several priority mixes and precedence strategies as a function of the number of dedicated circuit switching voice slots. Again, the moving boundary slot allocation policy offers significant advantages over the fixed boundary performance; for $\alpha = .3$, when S=30, the moving boundary is 2.5 ms. less for low priority traffic and .55 ms. less for the high priority traffic, both under precedence ordering 1. However, because the offered voice load is high relative to the number of slots dedicated to circuit switching, the individual slots are heavily utilized, and no delay threshold is attained until the number of voice slots allocated becomes large (Points X in the curve: S> 40).

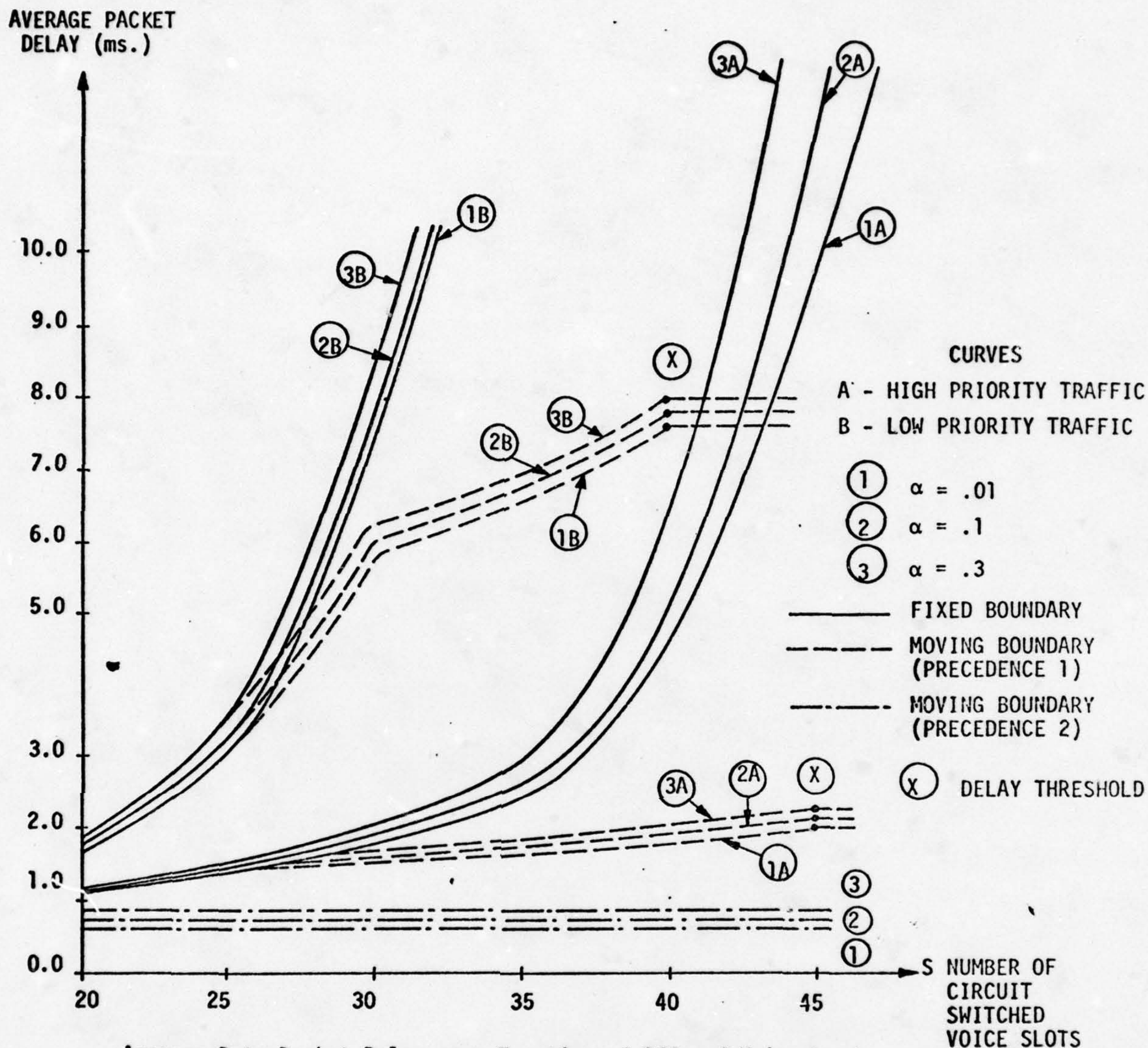
Figure 28 illustrates the blocking probability for low priority circuit-switched voice traffic as a function of the total offered voice load and the percentage of high priority traffic with a constant slot allocation, S=30. The effect of preemption



NOTE: Expansion shown for high priority traffic only

Expanded Diagram for Average Data Packet Delay as a Function of the Percentage of High Priority Traffic and Precedence Strategy

FIGURE 26: PACKET SWITCHING PERFORMANCE IN A PRIORITY BASED ENVIRONMENT FOR THE INTEGRATED CHANNEL

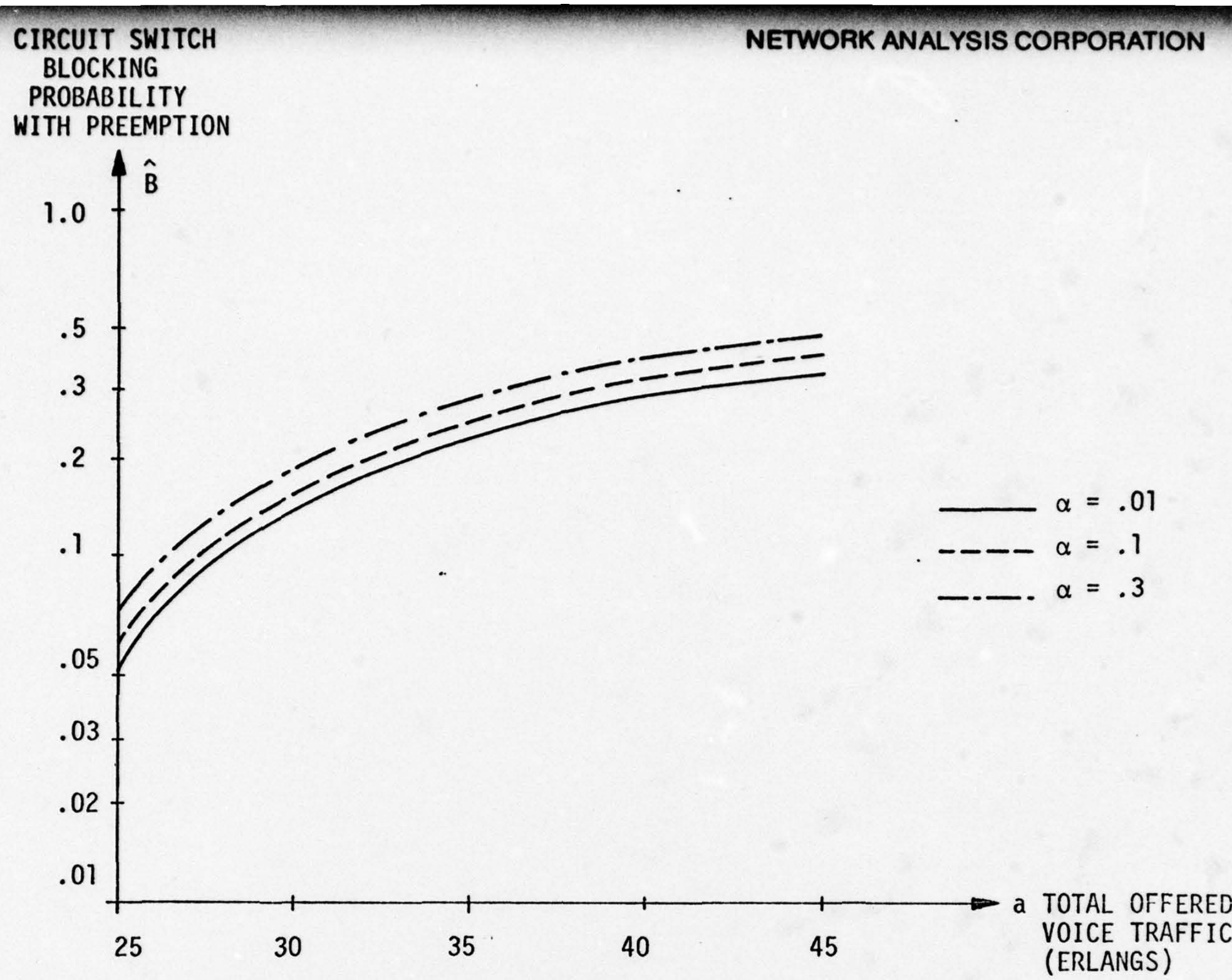


Average Data Packet Delay as a Function of Offered Voice Load and Precedence Strategy

FIXED PARAMETERS:

C = 1.544 Mbps	Θ = 250 PACKETS/SEC. (TOTAL)	VDR = 32 Kbps
b = 10 ms.	P = 1000 BITS	λ = 30 CALLS/ MINUTE (TOTAL)
	h = 1 MINUTE	

FIGURE 27: PACKET SWITCHING PERFORMANCE IN A PRIORITY BASED ENVIRONMENT FOR THE INTEGRATED CHANNEL



NOTE 1: The blocking probability for priority 2 voice traffic is augmented in the figure by formally including the impact of preemption (i.e., a low priority call can be blocked not only at its arrival but during transmission).

NOTE 2: For all cases, the high priority traffic blocking probability was negligible ($<10^{-4}$).

FIXED PARAMETERS:

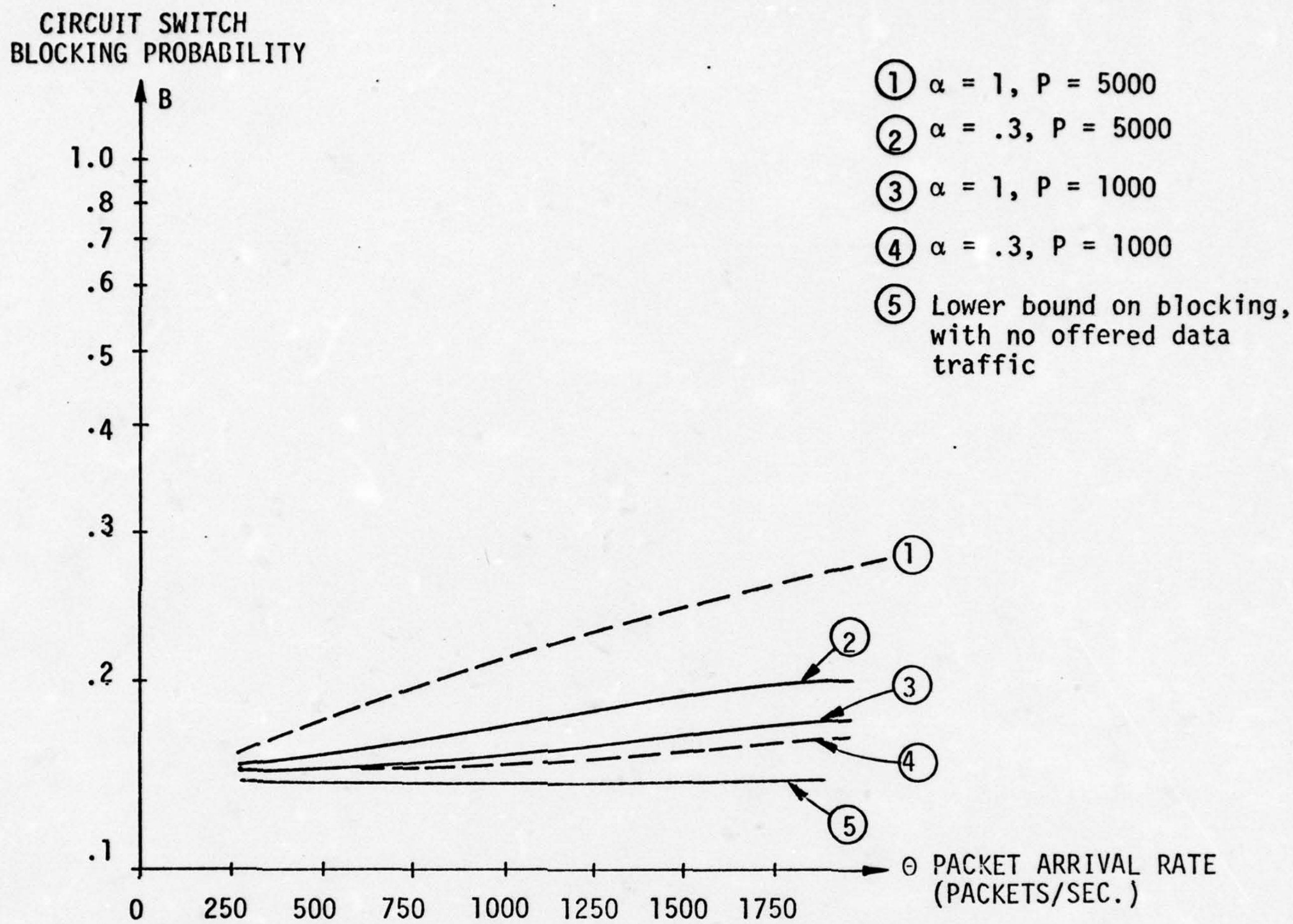
C = 1.544 Mbps	VDR = 32 Kbps	Θ(TOTAL) = 250 PACKETS/SEC.
b = 10 msec.	h = 1 MINUTE	S = 30
	P = 1000 BITS	

Blocking Probability as a Function of Offered Voice Load (With Preemption)

FIGURE 28: CIRCUIT SWITCHING PERFORMANCE IN A PRIORITY-BASED ENVIRONMENT FOR THE INTEGRATED CHANNEL

has been incorporated into our analytic models. Clearly, the low priority traffic blocking probability (slot unavailability at the instant a call arrives) is identical for any value of α (percentage of high priority voice traffic). However, the compound measure of system performance for low priority operation includes both the call blocking probability (connection denied at the moment of arrival) and the call preemption probability (connection broken for a low priority call already in progress). In fact, an increasing amount of high priority traffic (α=.01 → .1 →.3) results in a higher probability of call preemption, and consequently, a greater compound blocking probability as shown in Figure 28, ($\hat{B}$=.24 →.27 →.295 for a=35E).

Finally, the impact of data traffic on the low priority circuit-switching calls under precedence ordering 2 is shown in Figure 29. For the range of parameters previously studied, the absolute increase in blocking probability is relatively small. Only by allowing the percentage of high priority traffic to become extremely high ($\alpha \cong 1$), with a high data packet intensity ($\theta >$ 1000 packets/sec.) and larger packets (P=5000 bits), does the degradation to system performance become appreciable. For α =.3, P=5000 bits, the circuit traffic blocking probability increases from B=.168 when θ=250 packets/second to B=.196 when θ = 1750 packets/second.



Circuit Switch Blocking Probability as a Function of Packet Arrival Rate under Precedence Strategy 2 for Low Priority Traffic

FIXED PARAMETERS:

C = 1.544 Mbps	λ = 25 CALLS/MIN. (TOTAL)
b = 10 msec.	h = 1 MINUTE
S = 25	VDR = 32 Kbps

FIGURE 29: CIRCUIT SWITCHING PERFORMANCE IN A PRIORITY-BASED ENVIRONMENT FOR THE INTEGRATED CHANNEL

2.6 CONCLUSIONS

Analytical models for integrated switching links were developed and studies of integrated link performance were performed, as a function of input traffic, slot allocation, and slot management strategies. The performance models determine the probability of blocking for circuit switched traffic and the average delay for packet switched traffic on an integrated link. Special emphasis was placed on simplicity of the models to enable extension to network design, the incorporation of design parameters, such as, packet size and voice digitization rate, as well as the capability to address voice and data input traffic with priorities.

Parametric studies were conducted with respect to

- Channel performance as a function of traffic mix and slot allocation for both fixed and moving boundary policies.

- Channel performance under different slot assignment techniques such as:

  - Proportional (as a function of each traffic type).

  - Delay constrained (guaranteed level of performance for data).

  - Blocking constrained (guaranteed level of performance for voice).

- Impact on Blocking Probability and Delay of the following design variables under both the moving and fixed boundary slot allocation policies:

- Voice Digitization Rate.
- Speech Interpolation Capability.
- Packet Length.

- Data traffic with two different packet sizes, with and without priorities (this addresses the problem of integrating circuit switched signaling messages with regular data traffic).

- Channel Performance in a Priority Based Environment with respect to the:
  - Priority Mixture (% of high priority traffic).
  - Slot Allocation Policy (Fixed or Moving Boundary).
  - Precedence Strategy (based on priority class or traffic type).

Among the many interesting results that have emerged from the extensive experimentation, the following observations and conclusions are noteworthy concerning the performance of various slot allocation policies and slot assignment techniques:

- The moving boundary slot allocation policy is always to be preferred over the fixed boundary, not only for its obviously superior packet delay performance, but also because of an inherent "robustness", which enables channel performance to remain

largely unaffected by a poor choice for the number of dedicated circuit switching slots. In fact, it was observed that for fixed levels of voice loading, a delay threshold is ultimately reached, at which no additional degradation in service to data traffic can occur, independent of the number of dedicated circuit switching slots.

- Of the three slot assignment techniques (proportional, delay or blocking constrained) which were investigated, no particular one is uniformly superior. An integral point emphasized in the Appendix as well as by the experimental results was the greater efficiency of channel operation under increased traffic load when a blocking constraint rather than a delay constraint or proportional slot assignment technique is employed under the moveable boundary policy.

- The relationship between voice digitization rate and packet length was found to be of importance. Apart from the advantages offered by a lower VDR, if the packet length is chosen to be far in excess of the voice slot size, the performance improvement provided by the moving boundary policy may be marginal. This is, however, offset by the increased number of data slots that became available as a result of the lower total voice requirement of frame capacity. The choice of packet length is influenced, therefore, by header size, message length, frame duration and voice slot size.

Finally, it is noted that the issues of fixed vs. moveable boundary, VDR, data message size distribution, and the blocking probability and average packet delay for which the network is engineered, are all interrelated and subject to cost tradeoff resolutions. We offer the following conclusions:

- If a link is engineered for a relatively high blocking probability (e.g., "high usage" links in the AT & T network) then the advantages of a moveable boundary significantly diminish; thus the complexity and hence cost of the savings in channel capacity.

- A similar conclusion to that above results when the packet slot size is large compared to the voice slot size and no packet fragmentation is allowed. This may be the case when low VDR is used.

REFERENCES

[ANCKER 1961] Ancker, C. J. Jr., and A. V. Gafarian, "Queueing with Multiple Poisson Inputs and Exponential Service Times," Operations Research, 9, 1961, pp. 321 - 327.

[COBHAM, 1954] Cobham, A., "Priority Assignment in Waiting Line Problems," Operations Research, 1954, pp. 70-76 and 1955, pp. 547.

[COVIELLO, 1975] Coviello, G. and P. Vena, "Integration of Circuit/Packet Switching by a SENET (Slotted Envelope Network) Concept," National Telecommunications Conference, New Orleans, La., December 1975, pp. 42-12 to 42-17.

[DCA, 1975] Defense Communications Agency, "System Performance Specification for AUTODIN II, Phase I," November 1975.

[DCA, 1976] Defense Communications Agency, "Scenario of Defense Communications System (DCS) Common User Voice and Data Traffic - 1985 Estimate," part of the RFP Implications of Demand Assignment for Future Satellite Communication Systems, March, 1976.

[FISCHER, 1976] Fischer, M. J. and Harris, T. C., "A Model for Evaluating the Performance of an Integrated Circuit and Packet Switched Multiplex Structure," IEEE Transactions on Communications, February 1976, pp. 195-202.

REFERENCES CONT'D

[GEBERHARD, 1967] Geberhard, R. F., "A Queueing Process with Bi-Level Hysteresis Service Rate Control," Naval Research Logistics Quarterly, Vol. 14, 1967, pp. 43-54.

[GTE, 1975] GTE Sylvania, Inc., ESG. "SENET-DAX Interim Report," prepared for the Defense Communications Agency, December 10, 1975.

[MILLER, 1960] Miller, R. G., Jr., "Priority Queues", Ann. Math. Statist., 31, March 1960, pp. 86-103.

[MIYAHARA, 1975] Miyahara, H. et al., "A Comparative Evaluation of Switching Methods in Computer Communication Networks," International Conference on Communications, San Francisco, 1975, pp. 6-6 to 6-10.

[OCCHIOGROSSO, 1976] Occhiogrosso, B. J., "Performance of Integrated Communication Systems Using Shared Transmission Facilities and Conventional Switching Techniques," WM.15-76.03.R1, Network Analysis Corporation, July 28, 1976.

[RIORDAN, 1962] Riordan, J., Stochastic Service Systems, Wiley, New York, 1962.

[SAATY, 1961] Saaty, T. L. Elements of Queueing Theory with Applications, McGraw-Hill Book Co., Inc., New York, N. Y., 1961.

APPENDIX A

DERIVATION OF EQUATIONS FOR THE "PARTIAL AVAILABILITY MODEL"

In this appendix, we formally derive the average packet delay for the integrated switching model of Figure 9 based on the following state equation (see Eq. 24):

$$p_i = \sum_{K=0}^{i-1} \gamma_K \left((1-\sigma)p_{i-K} + \sigma p_{i+1-K}\right) + \gamma_i (p_0 + \sigma p_1) \qquad \text{(A.1)}$$

Defining the probability generating function:

$$P(z) = \sum_{i=0}^{\infty} p_i \; z^i \qquad \text{(A.2)}$$

and

$$\Omega(z) = \sum_{K=0}^{\infty} \gamma_K \; z^K \qquad \text{(A.3)}$$

We multiply each side of Eq. A.1 by z^i and taking the appropriate summation, one obtains:

$$p_i = \sum_{K=0}^{i-1} \gamma_K \left((1-\sigma)p_{i-K} + \sigma p_{i+1-K}\right) + \gamma_i (p_0 + p_1)$$

$$= \sum_{K=0}^{i} \gamma_K \left((1-\sigma)p_{i-K} + \sigma p_{i+1-K}\right) + \gamma_i \sigma p_0 \qquad \text{(A.4)}$$

$$\sum_{i=0}^{\infty} p_i z^i = \sigma p_0 \sum_{i=0}^{\infty} \gamma_i z^i + \sum_{i=0}^{\infty} z^i \sum_{K=0}^{i} \gamma_K \left((1-\sigma)p_{i-K} + \sigma p_{i+1-K}\right) \qquad \text{(A.5)}$$

or more simply:

$$P(z) = \sigma p_0 \Omega(z) + (1-\sigma) \sum_{i=0}^{\infty} \sum_{K=0}^{i} \gamma_K \, p_{i-K} z^i$$

$$+ \sigma \sum_{i=0}^{\infty} \sum_{K=0}^{i} \gamma_K p_{i+1-K} z^i \qquad \text{(A.6)}$$

Now using the generating function identity:

$$\sum_{i=0}^{\infty} \sum_{K=0}^{i} \gamma_K p_{i-K} z^i = \Omega(z)P(z) \qquad (A.7)$$

Equation (A.6) can be rewritten as:

$$P(z)=\sigma p_0 \; \Omega(z) + (1-\sigma)\; \Omega(z)P(z) + F \qquad (A.8)$$

where

$$F = \sigma \sum_{i=0}^{\infty} \sum_{K=0}^{i} \gamma_K \; p_{i+1-K} z^i \qquad (A.9)$$

and applying suitable algebraic simplification

$$F = \frac{\sigma}{z} \sum_{i=0}^{\infty} \sum_{K=0}^{i} \gamma_K \; p_{i+1-K} z^{i+1}$$

$$= \frac{\sigma}{z} \sum_{j=1}^{\infty} \sum_{K=0}^{j-1} \gamma_K \; p_{j-K} z^j \quad ,\text{with } j=i+1 \qquad (A.10)$$

$$F = \frac{\sigma}{z} \sum_{j=1}^{\infty} \left(\sum_{K=0}^{j} \gamma_K p_{j-K} z^j - \gamma_j p_0 z^j \right) \qquad (A.11)$$

$$F = \frac{\sigma}{z} \left(\sum_{j=0}^{\infty} \left(\sum_{K=0}^{j} \gamma_K p_{j-K} z^j - \gamma_j p_0 z^j \right) - \gamma_0 p_0 + \gamma_0 p_0 \right) \qquad (A.12)$$

and finally

$$F = \frac{\sigma}{z} \; (\Omega(z)P(z) - p_0 \Omega(z) \qquad (A.13)$$

Performing the required algebra in Eq. (A.8) we obtain:

$$P(z) = \frac{\sigma p_0 \Omega(z)(1-\frac{1}{z})}{1+\Omega(z)(\sigma(1-\frac{1}{z})-1)}$$

$$= \frac{p_0 \Omega(z)(z-1)}{z(1+\sigma\Omega(z)-\Omega(z)) - \sigma\Omega(z)} \tag{A.14}$$

which yields the generating function of Eq. (25)

$$P(z) = \frac{\sigma p_0 \; (z-1)}{z(\sigma+ \frac{1}{\Omega(z)} -1)-\sigma} \tag{A.15}$$

The probability of no packets in the system is obtained via:

$$P(z) \Big|_{z=1} = 1 \tag{A.16}$$

If the packet arrival process is assumed to be Poisson then $\Omega(z)$ is given as:

$$\Omega(z) = e^{-x} \; e^{xz} \tag{A.17}$$

where x is the mean number of packets which arrive in a single packet transmission time t,

$$x = \theta t \tag{A.18}$$

Applying L'Hospital's Rule to Eq. (A.15)

$$\lim_{z\to 1} P(z) = \lim_{z\to 1} \frac{\sigma p_0}{\sigma+ e^x e^{-xz} - 1 - xze^x e^{-xz}} = 1 \tag{A.19}$$

gives:

$$p_0 = 1 - \frac{x}{\sigma} = 1 - \frac{\theta t}{\sigma} \tag{A.20}$$

The average number of packets in the integrated switching system is derived from:

$$L = \left. \frac{\partial P(z)}{\partial z} \right|_{z = 1} \tag{A.21}$$

Two applications of L'Hospital's rule are required and omitting the algebraic tedium the end result becomes:

$$L = \frac{x(2-x)}{2(\sigma-x)} = \frac{\Theta t(2-\Theta t)}{2(\sigma-\Theta t)} \tag{A.22}$$

and finally from Little's formula, the average packet delay is obtained as:

$$D = \frac{t(2-\Theta t)}{2(\sigma-\Theta t)} \tag{A.23}$$

The applicability of the preceding formulation is limited for general (non-Poisson) packet arrival processes since the probability of K arrivals in a data slot interval depends not only on the length of the slot, but also the time since the last arrival occurred (often referred to as the residual lifetime). The Poisson arrival process possesses a memoryless interarrival time distribution so that the aforementioned dependency does not arise. The interslot dependence can be ignored if one assumes that the arrival process is "restarted" at the beginning of each data slot (the so-called queue with "synchronized" arrivals); this assumption preserves the direct applicability of Equation (A.14).

APPENDIX B

DERIVATION OF BLOCKING IN THE CASE OF SPEECH INTERPOLATION

In this appendix, we verify the result for the circuit switched voice blocking probability given in Eq. (36). Denoting y as the speech activity factor, then given j total conversations in progress, the probability that x are active, is assumed to be binominally distributed as:

$$\binom{j}{x} y^{x} (1-y)^{j-x} \qquad \text{(B.1)}$$

Assuming that the presence of all inactive conversations is simply recorded in the switch memory (infinite capacity), the probability that there are x active conversations, independent of the total number (active and inactive), is:

$$P_x = K \sum_{j=x}^{\infty} \binom{j}{x} y^{x} (1-y)^{j-x} \left\{ \frac{e^{-a} a^{j}}{j!} \right\} \qquad \text{(B.2)}$$

where K is a multiplicative constant and the bracketed term accounts for the infinite buffer capacity, with offered voice load a. Equation (B.2) reduces to:

$$P_x = K \frac{y^{x} a^{x} e^{-a}}{x!} \left(\sum_{j=x}^{\infty} \frac{a^{j-x} (1-y)^{j-x}}{(j-x)!} \right) \qquad \text{(B.3)}$$

The summation is the Taylor form for the exponential function, hence we obtain:

$$P_x = K \frac{y^{x} a^{x} e^{-a}}{x!} \left(e^{a(1-y)} \right) = K \frac{(ay)^{x}}{x!} e^{-ay} \qquad \text{(B.4)}$$

Since at most S conversations can be active for a given frame, the following probability conversation holds:

$$\sum_{x=0}^{S} P_x = 1 \quad . \qquad \text{(B.5)}$$

Solving for K, one finds:

$$K = \frac{e^{ay}}{\sum_{i=0}^{S} \frac{(ay)^i}{i!}} \qquad (B.6)$$

Thus, the probability of x active connections becomes:

$$P_x = \frac{\frac{(ay)^x}{x!}}{\sum_{i=0}^{S} \frac{(ay)^i}{i!}} \Pi = \Pi(a,x,S) \qquad (B.7)$$

APPENDIX C

SLOT MANAGEMENT SCHEMES

In this short Appendix, we wish to analytically highlight some relevant differences between the various slot assignment techniques:

- Proportional.
- Allocation under a fixed blocking constraint.
- Allocation under a fixed delay constraint.

and more fundamentally emphasize an important difference between facilities managed on a delay basis or a loss basis.

Recalling the earlier definitions,

S = Number of voice slots.

N = Number of data slots.

a = Offered voice traffic (erlangs).

θ = Packet Arrival Rate (packets/second).

t = Packet transmission time (seconds).

We now compare the various assignment techniques. Trivially, for all systems, if one fixes the performance at some specified constraint value, the channel capacity required to support an increasing amount of traffic must also increase, i.e.:

$$\frac{\partial N}{\partial \theta} > 0 \quad \text{(for fixed value of average delay)} \qquad \text{(C.1)}$$

$$\frac{\partial S}{\partial a} > 0 \quad \text{(for fixed level of blocking probability)} \qquad \text{(C.2)}$$

with fixed packet length and voice digitization rate. Under the ideal proportional capacity assignment:

$$N = K_1\theta \text{ , or } S = K_2 a \qquad \text{(C.3)}$$

for certain constants K_1, K_2. More significantly, the rate at which additional capacity must be acquired is constant; hence, no second-order penalty is incurred, nor economy of scale gained for capacity acquisition as traffic grows under the proportional assignment technique or:

$$\frac{\partial^2 N}{\partial \theta^2} = 0 = \frac{\partial^2 S}{\partial a^2} \qquad \text{(C.4)}$$

In this respect, the allocation of slots based on a delay constraint for the packet data traffic is identical to the proportional capacity assignment, i.e.,

$$\frac{\partial^2 N}{\partial \theta^2} = 0 \qquad \text{(C.5)}$$

as we now verify.

For the integrated switching channel the average packet delay is obtained in Section 2.3:

$$D = \frac{t}{2} \frac{(2-\theta t)}{(\sigma-\theta t)} \qquad \text{(C.6)}$$

Under a fixed delay constraint, the following relationship is implicitly defined:

$$\sigma = \frac{t}{D} + \theta\left(t-\frac{t^2}{2D}\right) \qquad \text{(C.7)}$$

In addition, the channel availability is by definition

$$\sigma = \frac{N+\phi_1}{N+\phi_2} \tag{C.8}$$

for constants ϕ_1, ϕ_2: $\phi_1 = [S/I]$, $\phi_2 = [\frac{S-\varepsilon}{I}]$:

$$N = \frac{\phi_1-\sigma\phi_2}{\sigma-1} = \frac{\phi_1-\phi_2}{\sigma-1} - 1 \tag{C.9}$$

Hence, straightforward differentiation reveals that:

$$\frac{\partial N}{\partial \Theta} = \frac{\partial N}{\partial \sigma} \frac{\partial \sigma}{\partial \Theta} = \left(\frac{\phi_2-\phi_1}{(\sigma-1)^2}\right)\left(t- \frac{t^2}{2D}\right) \tag{C.10}$$

Based on previous results, ϕ_2 ϕ_1 and since the minimum value D can ever achieve is simply the packet transmission time t, all terms in Equation (C.10) are positive and thus $\frac{\partial N}{\partial \Theta} > 0$. Now:

$$\frac{\partial^2 N}{\partial \Theta^2} = \frac{\partial N}{\partial \sigma} \frac{\partial^2 \sigma}{\partial \Theta^2} + \frac{\partial \sigma}{\partial \Theta} \frac{\partial}{\partial \Theta} \left(\frac{\partial N}{\partial \sigma}\right) \tag{C.11}$$

and clearly the second term of each product in the sum is identically zero, hence Equation (C.11) has been verified.

The operation of the channel slot assignment technique under a blocking constraint differs significantly from both the proportional technique and delay based strategy. Specifically, under constant loss (Erlang B equation):

$$\frac{\partial^2 S}{\partial a^2} < 0 \tag{C.12}$$

which implies that for a greater amount of traffic, less than the corresponding proportional number of slots must be added to the frame to maintain the identical grade of service, or alternatively individual slots in a larger circuit switching group operate more efficiently for fixed levels of blocking.

Hence, a certain performance economy of scale can be gained; this is quite advantageous, therefore, if a loss-based constraint is employed to assign slot capacity and the increased traffic is due primarily to voice. We omit here, for the sake of brevity, a proof of Equation (C.12). In an earlier work [OCCHIOGROSSO, 1976], the convexity of L (N, b) with respect to N was derived from which Equation (C.12) can be readily obtained (L(N,b) is the amount of traffic which can be carried on N trunks at a fixed level of loss, b). L(N,b) 's convexity implies that N (L,b) (number of trunks required to support traffic L, at fixed loss b) is concave with respect to L from which the negativity of the second derivative $\partial^2 S/\partial a^2$ follows.

CHAPTER 3

DESIGN OF INTEGRATED SWITCHING NETWORKS

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DESIGN OF INTEGRATED SWITCHING NETWORKS

3.1. INTRODUCTION

The design of integrated switched networks is addressed in this chapter. The traffic requirements offered to an integrated network includes traffic which is served by a circuit switched concept and traffic which is transported in a store-and-forward manner, using the packet switching concept. Furthermore, the circuit switched traffic requirements generate signaling messages whose functions are the establishment of end-to-end circuits and disconnection of same. The signaling messages must be transported in a store-and-forward manner and can be accommodated by the packet switched subnet.

The notion of circuit switching in the integrated network may not coincide with the classical notion, in that a physical end-to-end path may not exist while communication takes place in the circuit switched mode, and the traversing of a switch is not entirely transparent. The exact operation of the circuit switched concept would depend on the switching and transmission technology used to implement it. The general circuit switching notion as conceived here is one in which end-to-end switching and transmission facilities are "reserved" for a pair engaged in communication. Information communicated using this mode may be stored and forwarded when traversing a switching node. In contrast to the packet switched mode, however, the delay of the circuit switched information (during the communication period) when traversing a switch is virtually constant idependent on switch load; the outgoing link is a priori given. Furthermore, the circuit switched notion as used here is extendable to the case in which one takes advantage of idle (silent) periods in the circuit switched connection to transmit information from other circuit switched connections, signaling messages, or store and forward data. The above discussion implies a circuit

switched notion which is similar to a "reservation" approach, and in which the network control programs rather than the end users manage and control the end-to-end dedicated resources.

The notion of an integrated switched network assumed here is one in which the switching and transmission facilities are dynamically shared between traffic using the circuit switched mode and traffic using the packet switched mode. That is, the capacity of an ongoing link can be dedicated to a circuit switched connection at one instant and carry store and forward traffic at another instant. Similarly, the switching node contains all programs and functions needed to perform either circuit switching or packet switching and the central processor (or processors) is shared by all functions; its "instantaneous" load depends on the mix of traffic requirements, the priorities, etc., at the particular time.

The design of integrated networks encompases all the elements (subproblems) of circuit switched network design and packet switched network design. In addition, it includes functions which evolve from the dynamic sharing of resources by the two modes. Integrated network design has not been addressed in the literature to date. The integrated network is a novel concept for satisfying voice and data communication requirements, which may be composed of diverse application types and traffic characteristics. This chapter extends previous results of circuit and packet switched network design by addressing integrated circuit/packet switched network design. Briefly stated, the problem of integrated network design required the determination of minimum cost network resources (nodes, links, capacities) which satisfies average end-to-end delay for packet switched traffic, end-to-end loss probability for circuit switched traffic, average end-to-end delay for circuit connection set up, and reliability constraints. Similarly to circuit or packet switched network design, it includes the subproblems of routing, capacity assignment, and topological design.

Design variables, performance measures and design constraints for integrated networks are presented in Section 3.2. Section 3.3 addresses routing issues for circuit switched traffic; a routing approach for network operation which is based on estimates of probability of completing a connection and expected length of the path is introduced, and the derivation of the signaling traffic requirement are presented. The problems of capacity assignment and topological design are addressed in Section 3.4. Two options for link capacity assignment are presented and implemented in the program; the so called fixed and movable boundaries between link capacities for circuit switched and packet switched traffic. The analytical formulae, which give the average link packet delay, were developed in Chapter 2. In this chapter the synthesis problem is solved. An iterative optimization procedure which derives the set of link capacities satisfying the end-to-end average packet delays is presented.

Section 3.5 describes the computer program developed for integrated network design. At present the program designs the circuit switched subnet, then determines the link capacities for the signaling traffic and finally determines the total set of link capacities which satisfy grade of service requirements for all traffic types. The program is modular and enables easy extensions or modifications. The appendices contain derivation of formula presented in the body of the chapter and a detailed description of the integrated network design programs.

3.2. DESIGN VARIABLES, PERFORMANCE MEASURES, CONSTRAINTS

In this section we discuss the design variables, performance measures and constraints in the integrated network design problem.

3.2.1 Design Variables

1. Topology: Given by nodes (switches) and links (channels), and their interconnection.

2. Switch: The relevant parameters:

   - Capacity: expressed in terms of the number of communication links it can support, rate of circuit switching, rate of packet switching (or packet throughput).

   - Complexity: The mix of packet/circuit traffic it can handle, the handling of signaling traffic, frame management strategy.

   - Cost of the switch: Function of capacity and complexity.

3. Link: The relevant parameters are:

   - Capacity of the link.

   - Frame duration and management.

   - Boundary partition for circuit/packet slots in the frame.

   - Cost of the link, determined by the capacity and the mileage of the link.

4. Routing Policies: This issue will be discussed in more detail in Section 3. In the design phase, usually fixed routing policies are assumed because they are easier to analyze. In actual network operation, either fixed or adaptive routing can be used.

5. Traffic Requirement: For each pair of nodes (i,j), it includes the:

   - Average packet rate.

   - Average circuit offered load in Erlangs.

   - Average call holding time (for deducting the signaling requirements).

   In a more general context, the traffic requirements are only specified in terms of each traffic class (voice, interactive data, bulk file transfer, facsimile, etc.); the choice of the switching method for each traffic class becomes a design variable. (For example, it may be effective to use one-way circuit routes for file transfer to reduce the transmission overhead.)

6. Priorities: Voice and Data priorities and presedence ordering will exist in an integrated traffic DOD environment. The present program for integrated network design is capable of designing the store-and-forward subnet for

multiple priority traffic. In the future it will be extended to design circuit switched subnets for priority traffic.

7. Channel Flow: There are three types of flows on an integrated channel: circuit flow (i.e., circuit carried load), signaling flow, and the regular packet flow.

8. Other parameters used in network design are:

   - Voice digitization rate.

   - Packet size.

   - Overheads.

3.2.2 Performance Measures and Constraints

3.2.2.1 Packet Requirement Measures

The packet-switching network performance is measured in terms of source to destination packet delay. A commonly used performance measure in network design is the average end-to-end delay (average over all requirement pairs). Other performance measures, such as maximum end-to-end delay, can also be used. However, mathematically, they are less tractable.

3.2.2.2 Signaling Requirement Measures

Similar to the regular packet requirements, a reasonable signaling requirement performance measure is the average end-to-end set-up delay. Only signaling packets related to circuit set up and disconnection are considered, hence, for each origination call

three different signaling packets are required: an Inquiry Packet from source to destination to reserve a path, a Response Packet from the destination back to the source to signal that a path has been established, and a Disconnect Packet from the destination to the source (or vice versa) to release the path when the call is completed. The average end-to-end set-up delay can be considered as the average over the total inquiry and response time per call, or the average over the total inquiry, response and disconnect time per call; the latter is relevant in network design in order to minimize "wasted" capacity from end of "conversation" to the time links are reusable.

3.2.2.3 Circuit Requirement Measures

The circuit-switching network performance is usually measured in terms of the percentage of the calls that are lost. The commonly used performance measure in network design is the link blocking probability (e.g., AT&T telephone network links on the final route). The link measures are design-oriented, not user-oriented. However, the end-to-end loss probability measure seems mathematically intractible. In Section 3 we suggest two other alternative measures that can be used in the network design: the average link blocking probability and the average "end-to-end path" blocking probability. These measures reflect more directly the end-to-end loss than the link blocking measure.

3.2.2.4 Throughput Level

For the packet requirements, the base throughput γ_p is given by:

$$\gamma_p = \sum_{i,j} \gamma_{p,i,j} \qquad (1)$$

where $\gamma_{p,i,j}$ is the average packet requirements (packets/sec) from i to j. The base throughputs for signaling requirements and the circuit flow requirement are defined similarly.

Following the approach used in the packet-switched network design, we assume that in the network optimization the requirement pattern always remains fixed, whereas the total throughput level may be scaled up or down. This assumption is reasonable when adaptive routing (adapt to traffic pattern) is used.

3.2.2.5 Network Cost

The total backbone network cost is given by the cost of the switches and the cost of the communication channels. As mentioned before, the switch cost is a function of the switch capacity and the switch complexity, and the link cost is a function of the link capacity and mileage. Due to the economy of scale, it can be assumed that the link cost is a concave function of the capacity. The switch cost, on the other hand, usually varies at least linearly with the capacity. For a highly modularized switch (e.g., pluribus IMP), the linear cost model appears to be good. For less modular switch architectures (e.g., a cluster of identical processors) the cost tends to increase faster than capacity.

3.2.2.6 Flow Constraints

The average regular packet flows (and signaling packet flows) in links can be regarded as satisfying the multicommodity flow constraints with respect to the requirements $(\gamma_{p,s,t})_{s,t}$, i.e., if we let $f_{p,i,j}^{(s,t)}$ be the average regular packet flow on link (i,j) due to the requirement (s,t), then

(i) for each node i and requirement pair (s,t),

$$\sum_{j=1}^{NN} f_{p,j,i}^{(s,t)} - \sum_{j=1}^{NN} f_{p,i,j}^{(s,t)} = \begin{cases} -\gamma_{p,s,t} & \text{if } i = s, \\ 0 & \text{if } i \neq s,t, \\ \gamma_{p,s,t} & \text{if } i = t. \end{cases} \qquad (2)$$

(where NN is the total number of nodes),

(ii) for each link (i,j) and requirement pair (s,t),

$$f_{p,i,j}^{(s,t)} \geq 0$$

Note that the multicommodity flow constraint is satisfied only for the average flows. It does not hold for the instantaneous flows. Moreover, appropriate overhead factors must be taken into account. In Section 3, we show that the instantaneous circuit link carried loads approximately satisfy the multicommodity flow constraints.

It is obvious that capacity constraint must hold for the total instantaneous flow on each link, i.e., the sum of the regular packet flow (including overhead, etc), the signaling packet flow, and the circuit carried load on a link must be no larger than the total capacity of the link.

3.2.2.7 Connectivity Constraints

The backbone network is usually required to be 2-connected. Moreover, depending upon the application, there may be more restrictive constraints. For example, the nodes may require to have degree less than some fixed number because of a limitation on the switch ports.

3.3 CIRCUIT SWITCHED SUBNET: ROUTING AND DESIGN ISSUES

In this section we discuss several circuit-switched routing related issues. First we consider the routing strategies, and propose an adaptive circuit routing scheme taking advantage of the packet switching under the integrated envrionment. We then show that the instantaneous circuit link carried loads approximately satisfy the multicommodity flow constraints, under appropriate assumptions. We then examine the circuit requirement performance measures, and propose new design criteria: the average link blocking probability and the average end-to-end path blocking probability. These measures appear to be more appropriate than the usual criterion of link blocking probability. Finally, we derive the signaling requirement evaluation formula.

3.3.1 Routing Strategies

There are many advantages in integrating circuit-switching and packet-switching disciplines. For example, as was pointed out in [COVIELLO, 1975], using the SENET concept, the packet traffic can make use of the spare slots assigned for the circuit traffic, and thus achieve a substantial reduction in total capacity required. Also, the signaling packet traffic can share the capacity with regular packet traffic (with or without priority discipline) and thus achieve additional savings.

The standard circuit routing policy (either originating office control or progressive control) can be characterized as follows: For each pair of nodes (s,t), a set of routes (or next nodes) from s to t is specified in advance, and the routes (next nodes, respectively) are ordered as the first choice route, the second choice route, etc. Hence the routing policy is essentially fixed. The circuit signaling packet for a call always tries first to establish a path along the highest choice route. If this cannot be done, then it will try to seize a lower choice route. If no route is left,

the call is lost. Once a route is seized, it will be used for the entire duration of the call. The primary measure of effectiveness of the routing scheme is the average percentage of lost calls.

In contract, in packet-switching, the routing policies are usually adaptive or dynamic. There are no pre-determined routes for a requirement pair (s,t). At each node along the route for a packet, the next link is determined by choosing the one that gives the minimum estimated delay (or incremental delay) to the destination. Moreover, once the next link is selected, the packet will be queued for transmission along that link. Presumably only a very small fraction of packets will be lost. Hence, flow conservation can be assumed to hold. The primary measure of effectiveness of the routing scheme is the average end-to-end packet delay.

It is possible to employ a dynamic circuit routing policy, especially under the integrated environment (in this case, the routing information can be transmitted conveniently from node to node via the packet mode). One scheme can be as follows: At each node s in the network, store a table showing for each destination t the two (or more) outgoing links that correspond to the smallest expected blocking path, the second smallest expected blocking path, from s to t. Then, in establishing a path from s to t, each node a along the path will first attempt to reserve a channel on the link corresponding to the smallest expected blocking path. If this attempt fails, then it tries to reserve a channel on the link corresponding to the second smallest expected blocking path. If this attempt again fails, it is assumed that no path from a to t is accessible, and a signaling packet is sent back from a to s to signify the failure, and the call is lost. If a channel is successfully reserved on, say, the link (a,b), then the signaling reservation packet is sent from a to b via the usual packet routing scheme, and the channel

reservation process continues. In employing the above scheme, one must be on guard against the possibility of establishing excessively long paths. Since a circuit path is usually occupied for a long period of time (on the average of several minutes), the instantaneous best path may not be cost-effective. In order to prevent inadvertenly choosing a long path, one can store not only the blocking information, but also the expected path length information at each node.

Note that the routing discipline employed in the AT&T voice network is almost exactly the opposite; that is, the last route tried is usually the one with the least expected blocking. This can be justified because of the hierarchical nature of the AT&T network. Specifically, under the AT&T routing strategy, fewest expected number of links, and hence least amount of network resource, is used; moreover, the next call is most likely to be between some other node pairs in the network, and hence by using "local" resources first, the probability that the next call can be established is maximized. In our study, non-hierarchical network architecture is assumed, and hence exactly the opposite discipline is used to achieve the same objectives.

There are several other interesting routing issues in the integrated environment. For example, since the packet delay over a link is affected by the number of circuit calls using the same link, when the number of new circuit calls in the new frame interval exceeds a certain amount, it may be effective to reassign some of the packets waiting on the queue to other outgoing links.

3.3.2 Circuit Link Carried Loads as Multicommodity Flows

The average packet traffic on links can be approximately modeled as flows satisfying the multicommodity flow constraints, and hence the packet routing problem can be regarded as a flow problem. We would like to similarly model the circuit traffic. There are two types of circuit loads to a link: the circuit link

offered load and the circuit link carried load. The offered loads cannot be modeled as multicommodity flows except under very stringent assumptions, because a percentage of the offered load is usually blocked on each link. The circuit link carried load, on the other hand, can be regarded as approximately satisfying the multicommodity flow constraints, under appropriate (and realistic) assumptions, as will be shown below.

When a channel in a link is seized by a call, we can regard the link as having one unit of flow (contributed by that call) during the time interval in which the channel is seized. Now, the holding time for a call can be split into two parts. The first part consists of the call set-up time and the call disconnection time. The second part consists of the time interval during which the two ends engage in communication (including possibly many idle time intervals). During the first part only a partial path is usually occupied by the call, and hence the flow conservation law does not hold. During the second part, however, a complete end-to-end path is occupied by the call, and hence the flow conservation law holds. Under normal circumstances, we can assume that;

1. The percentage of lost calls is small (say, less than a few percent);

2. The average call set-up and disconnection time (which is usually less than a second) is much smaller than the average successful call holding time (which is usually on the order of several minutes).

It follows from 1 and 2 that the average time interval (averaged over successful and lost calls) for the second part (of the call holding time) is much larger than the average time interval for the first part. Consequently, at any instant, the link (i.e., the instantaneous link carried loads) in the entire network approximately satisfy the multicommodity flow constraints.

3.3.3 Circuit-Switched Network Performance Measures

The circuit-switched network performance is usually measured in terms of the percentage of lost calls. A good user-oriented measure is the average end-to-end loss probability (weighted average over all the requirement pairs) P_E. However, it is very difficult to obtain a closed-form (even approximate) formula for P_E except under very restrictive conditions. Consequently, more tractable measures are needed for the network design process.

Because of the difficulty to express the end-to-end probability of loss as a function of the probability of blocking on links, the link blocking probability is usually the measure used in circuit switched network design (inner loop). The end-to-end loss is computed via analysis (outer loop). This general approach is used in [KNEPLEY, 1973] for design of non-hierarchical circuit switched networks. In designing the AT&T hierarchical networks, the blocking probability on links on the "final route" is used as a design criterion. The *link blocking probability measure* indirectly bounds the average end-to-end loss probability: for a fixed routing policy, the smaller the link blocking probability, the smaller the average end-to-end loss probability. However, there is no explicit general formula relating the two.

Since the link blocking probability measure is only related to the average end-to-end loss indirectly, there is no reason to assume that it is the best criterion for the inner-loop design of the network optimization process. In fact, intuitively, the rigid constraint that all links must have roughly the same blocking makes it less effective with respect to cost-optimization. In this Section we present two alternative inner-loop design performance measures which appear to be more suitable for network optimization: the average link blocking probability, and the average "end-to-end path" blocking probability. One of the objectives in using these criteria is that they may allow a closer interplay between circuit and packet network design strategies under the integrated environment.

3.3.3.1 Average Link Blocking Probability

We define the average link blocking probabily P_{AL} for a network to be

$$P_{AL} = \frac{1}{NA} \sum_{\ell=1}^{NA} P_\ell \qquad (3)$$

where

NA = total number of links in the network,

P_ℓ = average blocking probability of link ℓ.

Using P_{AL} in design allows individual links to have different blocking probability.

In the following, under a specific routing scheme, we derive an approximate relationship between P_{AL} and P_E (the average end-to-end loss probability).

For illustrative purposes, we assume that the following progressive routing scheme is used: For each link ℓ on the primary path for a requirement pair (s,t), there is one alternate path from the initial node of ℓ to t which does not contain link ℓ. We call such an alternate path the alternate path branching from link ℓ, and denote it by $R_{s,t}(\ell)$. (Figure 1) Moreover, we assume that;

1. The system is in statistical equilibrium.

2. The occupancy distribution of the trunk groups throughout the network are statistically independent of each other.

3. No congestion is encountered at the switching node.

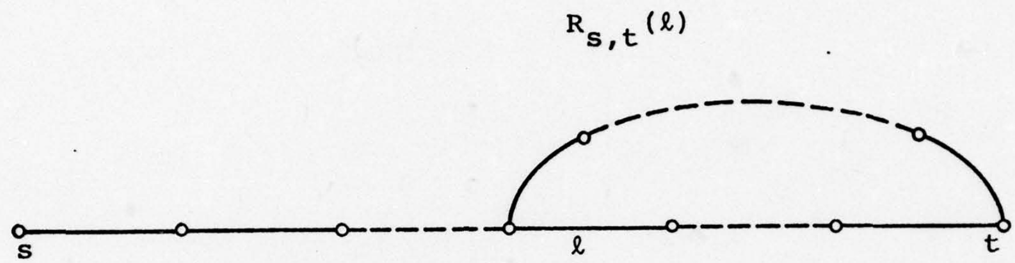


FIGURE 1: A PROGRESSIVE ROUTING SCHEME

4. All flow groups to a link encounter the same blocking probability.

5. The average blocking probability for a link is small (say, $\leq$ 5%).

6. The average number of links on an end-to-end path is small (say, $\leq$ 5).

Under these assumptions we show in Appendix A that

$$P_E \cong NL_P \times NL_A \times P_{AL}^2 \tag{4}$$

where NL_P and NL_A are the average number of links on a primary and an alternate path, respectively.

3.3.3.2 <u>Average End-to-End Path Blocking Probability</u>

Consider a call from s to t. The path R selected to transmit this call must be a *complete end-to-end* path from s to t (unless the call is lost). For the duration of the call, a circuit in each of the links in R is seized by the call, which corresponds to the carried load. We call such a path R the end-to-end path for the requirement (s,t).

Let P_{AR} be the average end-to-end path blocking probability. In Appendix B we show (under the assumptions of the previous section) that:

$$P_{AR} \cong \frac{1}{\gamma} \sum_{\ell=1}^{NA} f_\ell p_\ell \tag{5}$$

where γ is the total carried load in the network, f_ℓ is the carried load on link ℓ, and p_ℓ is the link blocking probability.

It is noted that this derivation is similar to Kleinrock's development of the average end-to-end delay formula [KLEINROCK, 1972]. Since the average end-to-end loss probability is usually required to be small (say, $\leq 2\%$), total carried load γ can be approximated by the total circuit offered load requirement.

Note that in the derivation of Equation (5), no assumption is made on the type of routing scheme used; thus the result is applicable under any routing strategy. (However, the routing stragegy does affect the flow assignment on the links.) Moreover, the derivation of Equation (5) can be modified slightly to take into account the more realistic assumption that different flow groups on the same link encounter different blocking probabilities, and the same formula, Equation (5), still holds.

Similar to P_{AL}, the relationship between P_E and P_{AR} can be crudely estimated to be

$$P_E \approx (P_{AR})^2 \tag{6}$$

for the progressive routing scheme described in Section 3.3.3.1.

Suppose the following type of "proportional" routing strategy is used: For each requirement pair (s,t), a set of end-to-end routes is given. When a call from s to t originates (at s), one of the routes, say R, is selected to carry the call to t, according to some probability distribution. If this call is blocked on R, then it is lost. Under this routing strategy, the average end-to-end path blocking probability is precisely the average end-to-end loss probability, and hence Equation (5) can be applied directly in the network design program to obtain the desired average end-to-end loss.

3.3.4 Signaling Traffic Requirements

In this subsection we discuss the generation of the signaling traffic requirement from the circuit offered loads, circuit carried

loads, and the link blocking probabilities. The signaling traffic load is small compared to the circuit traffic load. However, for the present study on the AUTOVON/AUTODIN database, the voice traffic is much larger than the regular data traffic, hence the signaling traffic is non-negligable when compared to the regular packet traffic under the integrated environment.

In deriving the signaling traffic requirements we consider the signaling messages used for connection set up and disconnection, according to the following model. For each circuit call, there are three types of associated signaling messages [MIYAHARA, 1975]:

1. Inquiry message;

2. Response message;

3. Disconnect message.

When a circuit call from s to t orginates at s, an inquiry packet is generated (at s). First, this packet tries to seize a free channel on the first choice link, say ℓ from s to t. If it succeeds, then the packet is sent from s to the node at the other end of link ℓ, either through a separate signaling network, or through the packet switched network; otherwise, the inquiry packet tries to seize a channel on the second choice link from s to t. In general, suppose the inquiry packet is at some intermediate node a (which is different from t), trying to seize a free channel on the k^{th} choice link, say ℓ, from a to t. If it succeeds, then the inquiry packet is sent from a to the node at the other end of link ℓ. Otherwise, there are two possibilities: if there is a (k+1)-th choice

link from a to t, then the inquiry packet tries to seize a free channel on this link and the path-searching operation proceeds. Otherwise, the inquiry packet assumes that no path can be established from a to t; a response packet is generated at node a and sent back to s to inform s of this fact (and thus the call is lost). Moreover, in this case, a disconnect packet is also generated at a, which travels back to s through each node on the partial path established from s to a, in order to free up the seized channels on this partial path. If the inquiry packet reaches t, then a circuit path has been successfully established. A response packet is then generated at t, and sent from t to s, to inform s to start the transmission. After the call is completed a disconnect packet is generated at t, which travels from t to s through each node on the circuit path used, in order to free up the seized channels. (Of course, it is possible to first send an inquiry packet from s to t, and then try to establish a path from s to t in a backward fashion using the response packet. However, the analysis is similar.)

In conclusion, for each call from s to t, a route R consisting of nodes $s_0 = s$, $s_1, \ldots, s_k$ is established, and either $s_k = t$, or the channel-seizing operation fails to proceed at s_k. The signaling requirements associated with this call is as follows: For each i=0, 1,...,k-1, there is one signaling packet requirement from s_i to s_{i+1} corresponding to the inquiry packet used in seizing a circuit in link (s_i, s_{i+1}), and one signaling packet requirement from s_{i+1} to s_i corresponding to the disconnect packet used in releasing the seized circuit of link (s_i, s_{i+1}). Also, there is one signaling packet requirement from t to s corresponding to the response packet used to inform s of the success or failure. This is illustrated in Figure 2.

The signaling traffic requirements can easily be computed according to the model described. Let $\ell = (a,b)$; we denote by:

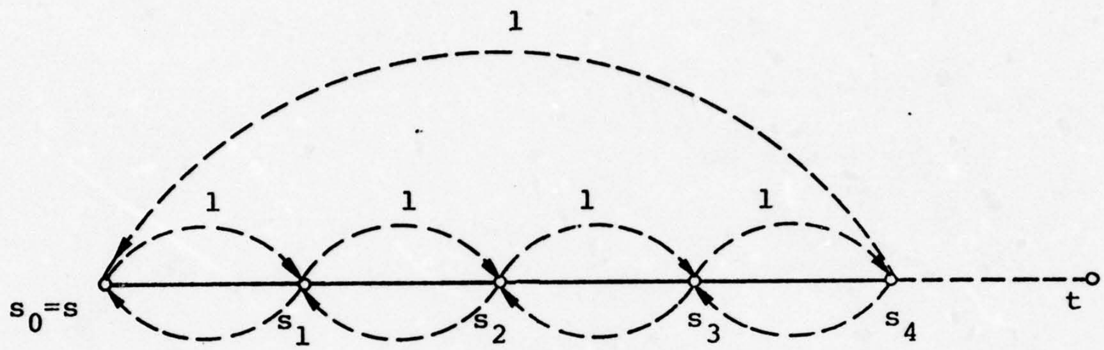


FIGURE 2: SIGNALING PACKET REQUIREMENTS CORRESPONDING TO ONE CALL

I_{ab} = Average inquiry packet requirements from a to b (packets/sec).

R_{ab} = Average response packet requirements from a to b (packets/sec).

D_{ab} = Average disconnect packet requirements from a to b (packet/sec).

γ_{ab} = Average offered load in Erlangs from a to b.

p_{ab} = Blocking probability on link (a,b).

q_{ab} = Rate of calls lost at a when unsuccessful in reserving circuit on link (a,b).

H = Average call holding time (sec).

Then:

$$\left.\begin{aligned} I_{ab} &= (1-p_{ab}) \quad \gamma_{ab}/H \\ D_{ab} &= I_{ab} \\ R_{ab} &= \gamma_{ba}/H + q_{ab} \end{aligned}\right\} \qquad (7)$$

Now we consider briefly the average signaling set-up delay. For each link ℓ, let

fI_ℓ = average number of inquiry packets/sec on link ℓ,

fR_ℓ = average number of response packets/sec on link ℓ,

fD_ℓ = average number of disconnect packets/sec on link ℓ,

T_ℓ = average delay of signaling packets on link ℓ.

γC = average total number of calls/sec (over all the requirement pairs) in the network.

$T_{I,R}$ = average signaling set-up round trip delay (inquiry + response),

$T_{I,R,D}$ = average signaling total transmission delay (inquiry + response + disconnect).

Similar to the derivation for Equation (3) we can show that;

$$T_{I,R} = \frac{1}{\gamma C} \sum_{\ell} (fI_{\ell} + fR_{\ell})T_{\ell}, \tag{8}$$

and

$$T_{I,R,D} = \frac{1}{\gamma C} \sum_{\ell} (fI_{\ell} + FR_{\ell} + fD_{\ell})T_{\ell}. \tag{9}$$

$T_{I,R}$ is a meaningful measure for the signaling set-up delay. However, a direct application of the extremal-flow routing program [NAC, 1976b] requires that the total flow on each link be considered as a single entity, hence, it is only applicable to $T_{I,R,D}$. This however is not a limitation since the disconnection delay must also be bounded to minimize the "wasted" capacity during this process.

3.4 ROUTING, CAPACITY ASSIGNMENT, AND TOPOLOGICAL DESIGN OF INTEGRATED NETWORKS

In this section we discuss the topological design methods for integrated backbone networks. The basic approach taken is that of merging the corresponding design methods for the circuit switched network and the packet switched network. The related network design variables and performance measures have been discussed in Section 3.2. Evaluation procedures for the signaling requirements, average end-to-end signaling delay, average end-to-end regular packet delay and the average end-to-to end circuit call loss probability are presented in the previous section.

The topological design problem for an integrated backbone network can be formulated as follows:

<u>Given</u>

- Nodal switch locations.
- (Circuit switched and packet switched) Traffic requirements.
- Cost vs. capacity function $D_i = d_i(C_i)$ for all potential links, where C_i is the capacity of link i.
- Cost vs. capacity function $F_i = f_i(S_i)$ for all nodes, where S_i is the capacity of switch i.
- Voice digitization rate.
- Signaling packet size.
- Average regular packet size/packet overhead.

<u>Find</u>

- A set of links $A = (\ell_i)_i$.
- Capacity C_i for each of the links in A.
- Capacity S_i for each of the nodes.

Which <u>minimizes</u> $D = \sum_{i=1}^{NA} d_i(C_i) + \sum_{i=1}^{NN} f_i(S_i)$

such that

- The traffic requirements are accommodated.
- The average circuit call loss constraint is met.
- The average end-to-end regular packet delay constraint is met.
- The average signaling delay constraint is met.
- Other appropriate constraints (e.g., 2-connectivity) are met.

The topological design of a distributed non-hierarchical communications network is a very complex and challenging problem. Some of the contributing factors to problem complexity are the combinatorial nature of the choice of topologies, the integral nature of the link capacities, the interactive nature of the circuit routing schemes, and the presence of the complex circuit call completion and packet response time requirements. As a consequence, the general problem can only be solved by approximation methods. Exact solution methods exist only for some well-defined special cases and subproblems. A general solution approach for the total problem is by interacting the following two subproblems: the topology modifi-

cation, and the flow and capacity assignment for a given topology. The selection of the best topology is inherently a very complex problem, and it appears likely that no polynomial time exact solution algorithm exist. (Even a simpler problem, the traveling-salesman problem, is known to be NP-complete.) The circuit routing and/or capacity assignment is again a very difficult problem, due to the inherent complex interaction of the circuit routes, especially in the non-hierarchical environment. The packet flow and/or capacity assignment problem, on the other hand, allows exact solution methods using mathematical programming and numerical analysis techniques, under appropriate concave continuous cost/capacity assumptions and (approximate) closed-form average end-to-end delay expressions [FRATTA, 1973].

In the following, we first outline a solution procedure for circuit network routing and capacity assignment. We then consider the integrated network link capacity assignment, given that the link circuit and packet flows are known, and present a solution method based on numerical techniques. Combining these solution procedures and their variations, and the known packet switching routing algorithm [CANTOR, 1974], we describe several heuristic solution approaches to the integrated network routing and capacity assignment problem. Finally we examine the existing solution techniques for the topology modification problem. By combining the solution procedures for these two subproblems, we have solution approaches for the total integrated network design problem.

In the following, we assume that the link costs are given by the power-law cost function: Namely, the cost of link i is;

$$D_i = (K_1+K_2 x d_i) \times C_i^{\alpha}, \tag{10}$$

where

d_i = Mileage of link i,

C_i = Capacity of link i (K bits/sec),

α = Power cost factor,

K_1 = Fixed cost coefficient,

K_2 = Mileage cost coefficient.

Using the power link cost model, the integrated link capacity assignment problem can be solved using standard numerical techniques (Section 3.4.2). Notice that the actual tariff structure in reality is approximated much better with a discrete link cost model. However, due to its combinatiorial complexity, this type of problem can only be solved using non-optimum heuristic methods. Consequently, the trade-off is between the capability of obtaining near-optimum solutions (usually as close to optimum as one desires) to the less realistic continuous link cost model, or cruder heuristic solutions to the more realistic discrete link cost model. In this study, we use the former approach.

3.4.1 Circuit Network Routing and Capacity Assignment

In this subsection we present a heuristic solution method for the Circuit network Routing and Capacity Assignment problem (CRCA). This approach is a variation of the circuit network design algorithm described [NAC, 1976b].

Given:

- Network topology (nodes, links)
- Circuit traffic requirements

- Average end-to-end loss objective P_E^* and tolerance Δ

Procedure

1. Initialize arc weights.
   Initialize P_L, the link blocking constraint.
   Initialize network cost COST, to ∞.

2. Find the shortest paths for the given arc weights. Use these paths as the primary circuit routes for the requirements.

3. Based on the same arc weights and the primary routes, find the alternate routes for the requirements.

4. Load the circuit requirements onto the primary and the alternate routes, using P_L as the proportion of overflow traffic for each link. (This step gives a rough estimate of the link flow, and is used in the next step for trunk sizing.)

5. Based on the link average offered load estimates f_ℓ obtained in Step 4 and the Erlang-B formula, find the number of circuits N_ℓ for each link such that N_ℓ is the smallest positive integer with $B(f_\ell, N_\ell) \leq P_L$. If f_ℓ is 0, then set N_ℓ to 0.

6. Based on the trunk sizes N_ℓ, and the primary and the alternate routes, load the requirements onto the links using the iterative loading scheme described in [NAC, 1976b]. Compute the average end-to-end loss P_E.

7. If $|P_E - P_E^*| > \Delta$, obtain a new estimate for P_L based on some effective updating scheme, and go to Step 4.

8. Calculate network cost $COST_2$.
   If $COST_2 < COST_1$, then

   a. $COST_1 \leftarrow COST_2$

   b. Update the arc weights, using the weighting functions suggested in [NAC, 1976b].

   c. Go to Step 2.

9. Output the network design associated with $COST_1$.

10. End.

3.4.2 Integrated Network Link Packet Capacity Assignment Problem

In this subsection we consider the Integrated network Packet link Capacity Assignment problem (IPCA), given the circuit and packet flows on each link. Formally stated:

Given:

- Nodes 1,2,...,NN

- Links 1,2,...,NA

- Average circuit carried load V_i in terms of Kb/s on each link i (V_i is the link circuit capacity if fixed boundary strategy is used), $i=1,\ldots,NA$.

- Average packet flow f_i (Kb/s) on each link i, $i=1,\ldots,NA$.

- Total packet traffic requirement γ_p.

- Link cost/capacity function

$$D_i = (K_1+K_2 x d_i) \times C_i^{\alpha},$$

for $i=1,\ldots,NA$.

- Average end-to-end delay constraint $\overline{T}$.

<u>Minimize</u>

$$\text{Total cost } D = \sum_{i=1}^{NA} D_i \tag{11}$$

<u>Such that</u>

$$\frac{1}{\gamma_p} \sum_{i=1}^{NA} f_i \, T_i \leq \overline{T}, \tag{12}$$

where T_i is given by (see Chapter 2)

$$T_i = \frac{P}{2C_i} \cdot \frac{2C_i-f_i}{C_i-V_i-f_i} \tag{13}$$

(P is the packet length, a constant).

Using the method of Lagrangian multipliers, the above problem is equivalent to;

$$\min G(\beta, C_1, \ldots, C_{NA}) = \sum_{i=1}^{NA} D_i + \beta\left[\frac{1}{\gamma_p} \sum f_i T_i - \bar{T}\right] \qquad (14)$$

The minimum occurs at the point where all the first partial derivatives vanish. Hence we obtain the following set of NA+1 equations with NA+1 unknowns $\beta, C_1, \ldots, C_{NA}$ must be solved.

$$4\alpha(K_1+K_2 d_i) C_i^{1+\alpha}(C_i - V_i - f_i)^2 = \frac{\beta P}{\gamma_p} f_i[(2C_i - f_i)^2 + f_i(2V_i + f_i)],$$

$$i=1,\ldots,NA \qquad (15)$$

$$\frac{P}{\gamma_p} \sum \frac{f_i}{2C_i} \frac{2C_i - f_i}{C_i - V_i - f_i} = \bar{T} \qquad (16)$$

Moreover, in order for the solutions to be meaningful, it must satisfy:

$$C_i \geq V_i + f_i, \qquad (17)$$

for i=1,...,NA.

An analytic solution for the above equations appears unlikely even in the case $\alpha=1$ (i.e., linear cost/capacity function). However, based on these equations we deduced some general functional behavior, relating β, C_i's and $\bar{T}$. From which efficient numerical solution methods are devised.

First, from Equation (15), we can express β in terms of C_i, for i=1,...,NA as:

$$\beta = \frac{4\alpha(K_i+K_2 d_i)\gamma_p}{f_i P} \frac{C_i^{1+\alpha}(C_i - V_i - f_i)^2}{[(2C_i - f_i)^2 + f_i(2V_i + f_i)]}, \qquad (18)$$

Consequently,

$$\frac{\partial\beta}{\partial C_i} = \frac{4\alpha(K_1+K_2d_i)\gamma_p}{f_iP} \quad \frac{\phi(C_i)}{[(2C_i-f_i)^2+f_i(2V_i+f_i)]^2}, \tag{19}$$

where

$$\phi(C_i) = [(1+\alpha)C_i^{\alpha}(C_i-v_i-f_i)^2 + 2C_i^{1+\alpha}(C_i-V_i-f_i)]x \tag{20}$$

$$[(2C_i-f_i)^2 + f_i(2V_i+f_i)] - [C_i^{1+\alpha}(C_i-V_i-f_i)^2 4(2C_i-f_i)]$$

Note that for $C_i>V_i+f_i$,

$$\phi(C_i) > 2C_i^{1+\alpha}(C_i-V_i-f_i)(2C_i-f_i)^2 - 4C_i^{1+\alpha}(C_i-V_i-f_i)^2(2C_i-f_i)$$

$$= 2C_i^{1+\alpha}(C_i-V_i-f_i)(2C_i-f_i)[(2C_i-f_i)-2(C_i-V_i-f_i)]$$

$$> 0. \tag{21}$$

We thus conclude that for i=1,...,NA, if $C_i>V_i + f_i$, then $\frac{\partial\beta}{\partial C_i} > 0$, i.e. for $C_i> V_i + f_i$, β is a monotonically increasing function of the C_i's.

Next, we consider the relationship between T and the C_i's. From Equation (16), for i=1,...,NA,

$$\frac{\partial T}{\partial C_i} = \frac{P}{\gamma_p} \frac{\partial}{\partial C_i}\left(\frac{f_i}{2C_i} \frac{2C_i-f_i}{C_i-V_i-f_i}\right)$$

$$= \frac{P}{\gamma_p}\left[\frac{-f_i}{2C_i{}^2}\left(2+\frac{2V_i+f_i}{C_i-V_i-f_i}\right) + \frac{-f_i}{2C_i} \frac{2V_i + f_i}{(C_i-V_i-f_i)^2}\right]. \tag{22}$$

Consequently, if $C_i > V_i + f_i$, them $\frac{\partial T}{\partial C_i} \leq 0$, i.e. if for i=1,...,NA, $C_i > V_i + f_i$, then T is a montonic decreasing function of the C_i's. We thus have the following result:

Theorem 1: If for i=1,...,NA, $C_i > V_i + f_i$, then T is a monotonic decreasing function of β.

Theorem 1 indicates that each T corresponds to a unique β (for meaningful values of the C_i's). Hence the desired β can be obtained from $\overline{T}$ by applying linear or quadratic functional evaluation techniques to $T = T(\beta)$ [HAMMING, 1962].

In order to solve for T for a given β, one must first solve for the C_i's for the given β using Equation (15). As will be shown below, if we express the function

$$F(C) = 4\alpha(K_i+K_2d)C^{1+\alpha}(C-V-f)^2 - \frac{\beta P}{\gamma_p} f[(2C-f)^2 + f(2V+f)] \tag{23}$$

as a function of $x = \frac{1}{C}$, then for fixed $\beta > 0$, the function

$$\begin{aligned} H(x) &= x^{3+\alpha}F(x) \\ &= 4\ (K_1+K_2d)((V+f)x-1)^2 - 2\ \frac{\beta P}{\gamma_p} f\ x^{1+\alpha}(2-2fx + f(V+f)x^2) \end{aligned} \tag{24}$$

is monotonically decreasing for $0 \leq x \leq \frac{1}{V+f}$, furthermore $H(x)<0$ for $x=\frac{1}{V+f}$, and $H(x)>0$ for $x=0$. Consequently, there exist a unique root x for H(x) in the range $(0, \frac{1}{V+f})$. Hence x can be solved for a given β using linear or quadratic functional evaluation techniques to Equation (24). The C_i's can then be obtained by setting $C_i = \frac{1}{x_i}$.

<u>Theorem 2:</u> For fixed $\beta>0$, the function H(x) given by Equation (24) has the following properties:

1. $H(x) > 0$ for $x=0$.

2. $H(x) < 0$ for $x=\frac{1}{V+f}$.

3. H(x) is monotonically decreasing for x in $(0, \frac{1}{V+f})$. Consequently, there exist a unique root for H(x) in the range $(0, \frac{1}{V+f})$.

<u>Proof</u>

$H(0) = 4\alpha(K_1+K_2d) > 0$, so property (1) holds.

$$H(\tfrac{1}{V+f}) = -2\frac{\beta P}{\gamma_p} f(\tfrac{1}{V+f})^{1+\alpha}[2-f(\tfrac{1}{V+f}) + f(V+f)(\tfrac{1}{V+f})^2]$$

$$= -4\frac{\beta P}{\gamma_p} f(\tfrac{1}{V+f})^{1+\alpha} < 0, \tag{25}$$

so property (2) also holds.

It remains to prove property (3). From Equation (24),

$$H'(x) = 8\alpha(K_1+K_2d)(V+f)[(V+f)x-1]$$

$$-2\frac{\beta P}{\gamma_p} f\ [(1+\alpha)x^{\alpha}(2-2fx+f(V+f)x^2) + x^{1+\alpha}(-2f+2f(V+f)x)]$$

$$= 8\alpha(K_1+K_2d)(V+f)[(V+f)x-1]$$

$$-2\frac{\beta P}{\gamma_p} fx^{\alpha}\ (2(1+\alpha)-2f(2+\alpha)x+(3+\alpha)f(V+f)x^2) \tag{26}$$

By assumption, $x<\frac{1}{V+f}$, so

$$8\alpha\ (K_1+K_2d)(V+f)[(V+f)x-1] < 0. \qquad (27)$$

Also, consider the quadratic equation

$$Q(x) = 2(1+\alpha)-2f(2+\alpha)x + (3+\alpha)f(V+f)x^2. \qquad (28)$$

The discriminant of this equation is;

$$\Delta = B^2-4AC = 4f^2(2+\alpha)^2 - 8(1+\alpha)(3+\alpha)\ f(V+f) < 0. \qquad (29)$$

Consequently, for any values of x,

$$Q(x)>0 \qquad (30)$$

Combining (27) with (30), we obtain that $H'(x)<0$, for x in $(0, \frac{1}{V+f})$ which is property (3).

Q.E.D.

Similar techniques can be applied to solve the capacity assignment problem if there is more than one class of packet switched traffic (e.g. mixing the signaling and the regular packet traffic); it has been implemented in the design program via traffic with priorities of which the above is a special case.

3.4.3 Solution Approaches for the Integrated Network Routing and Capacity Assignment Problem

Based on the solution methods given in Sections 3.4.1 and 3.4.2 for the CRCA and the IPCA problems, and the known packet switching routing algorithm [CANTOR, 1974], we can devise solution procedures for several other related network design subproblems. These procedures can then be combined to solve the Integrated network Routing and Capacity Assignment problem (IRCA). The procedures described in the following sections are implemented in the integrated network design program described in Section 3.5.

3.4.3.1 Integrated Network Packet Routing and Capacity Assignment Problem (IPRCA)

In this problem, given the topology, the link circuit carried load, the link circuit capacity, the packet traffic requirements, it is necessary to find the packet switching routes and the link capacities so that the network cost is minimized while satisfying the traffic requirements and the packet average end-to-end delay constraint. We outline a solution procedure for this problem below:

Procedure for IPRCA

1. Assign initial arc lengths.

2. Based on the arc lengths given, do shortest path routing. Use these routes as the initial packet switching routes.

3. Load the packet traffic requirements onto the links based on the routes obtaines in Step 2.

4. Based on the link circuit and packet flow, do capacity assignment, using the numerical procedure described in Section 3.4.2.

5. If the network cost fails to improve, then end; Output the routes and the capacity assignments.

6. Else, do optimum multi-path routing [CANTOR, 1974]. Go to 4.

3.4.3.2 Integrated Network Routing Problem (IRT)

In this problem, given the link circuit capacity $\{NCH(i)\}_i$ and the link total capacity $\{C(i)\}_i$, the circuit and packet traffic requirements, it is necessary to find the circuit, regular packet, and signaling packet routes so as to minimize the average circuit end-to-end loss P_E, the average packet end-to-end delay T_E and the average signaling delay S_E, while accommodating the traffic requirements.

A solution procedure for this problem is as follows:

IRT Procedure

1. Assign initial arc lengths (for the circuit requirements) (e.g., $\frac{1}{NCH(i)}$). $P_E(1) = 1$.

2. Generate shortest paths based on the arc lengths obtained. Use these paths as the circuit primary routes.

3. Generate alternate circuit routes.

4. Based on the trunk size NCH(i), and the primary and the alternate routes, load the circuit requirements on to the links. Obtain;

   a. The average link carried load V(i) for each link i,

   b. The average link blocking probability $P_L(i)$ for each link i,

   c. The average end-to-end loss P_E.

5. If $P_E < P_E(1)$, then

   a. Update arc lengths; use $(\partial P_L(i)/\partial V(i))_i$ as the new arc lengths.

   b. $P_E(1) \leftarrow P_E$.

   c. Go to Step 2.

6. Else, generate the signaling requirements, based on the circuit requirements and the circuit routes associated with $P_E(1)$, using the signaling requirement evaluation procedure given in Section 3.3.4.

7. Route the signaling requirements and the regular packet requirements so as to minimize T_E and S_E, using extension of the external-flow multipath packet routing procedure.

Since the signaling traffic is usually much smaller than the regular packet traffic, good approximations for the regular packet link delays can be obtained by modifying the regular packet flows to include also the signaling packet flows (and reducing the signaling flows to 0). Moreover, by assigning signaling packets higher priority over the regular packets, the signaling delay constraint will be satisfied. Consequently, the routing of the regular and signaling packet requirements can be achieved by considering it simply as the routing of only one class of packet requirements. This can be incorporated in the design procedure described by proper modifications of Steps 6 and 7.

3.4.3.3 Integrated Network Capacity Assignment Problem (ICA)

In this problem, given the circuit, packet and signaling routes, and the circuit and regular packet traffic requirements, it is desired to find the link circuit capacities $\{NCH(i)\}_i$ and the link total capacities $\{C(i)\}_i$, so as to minimize the total network cost while accommodating the traffic requirements, and satisfying the average circuit end-to-end loss constraint P_E^*, the average packet end-to-end delay constraint T_E^* and the average signaling delay constraint S_E^*.

A solution procedure for this problem is as follows:

ICA Procedure

1. Obtain initial link blocking bound P_L.

2. Load the circuit requirements onto the primary and the alternate routes, using P_L as the proportion of overflow traffic of each link.

3. Based on the link average offered load estimate f_ℓ obtained in Step 2 and P_L, do trunk sizing using the Erlang-B formula.

4. Load the traffic requirements onto the primary and the alternate routes, using the iterative loading scheme described [NAC, 1976b]. Obtain the average link carried load V_i for each link, and the average end-to-end loss P_E.

5. If $|P_E-P_E^*| > \Delta$, obtain a new estimate for P_L, and go to Step 2.

6. Else, based on the circuit traffic requirements and the average link carried loads, generate the signaling traffic requirements.

7. Load the signaling and regular packet flow onto the links based on the signaling and the regular packet routes given.

8. Do link capacity assignment based on extensions of the capacity assignment procedure described in Section 3.4.2.

Similar to the integrated routing problem, the integrated capacity assignment problem can be handled by merging the signaling and the regular packet traffic into one class, and disregarding the signaling delay constraint in the capacity assignment.

3.4.3.4 Integrated Network Routing and Capacity Assignment (IRCA)

Using the solution procedures for the subproblems of the integrated routing and capacity assignment problem, several alternative solution procedures for the IRCA can be constructed. Several design procedures can be constructed corresponding to the following two sets of options:

1. Signaling Transmission Options:

   a. Separate link capacities for the signaling traffic and the regular packet traffic.

   b. Shared capacity between signaling traffic and the regular packet traffic.

2. Design Procedure Options:

   a. Do the routing and capacity assignment problem for the circuit switched traffic and the packet switched traffic separately.

   b. Do the routing problem for the combined circuit and packet traffic. Do the capacity assignment problem for the combined circuit and packet traffic.

Options 1a and 2a give rise to simpler design procedures. However, they also produce less cost-effective designs. We outline the procedure which is currently implemented (see Section 3.5).

Solution Approach

1. Based on the circuit traffic requirement and the average end-to-end loss constraint, do the circuit switched routing and capacity assignment using the procedure described in Section 3.4.1.

2. Based on the circuit requirements and the circuit routes obtained in Step 1, calculate the signaling traffic requirements using the formulas derived in Section 3.3.4.

3. Apply the IPRCA procedure to the signaling requirements by setting $V_i = 0$ for all link i. Let the signaling capacity thus obtained be CS_i, for each link i.

4. Apply the IPRCA procedure to the regular packet requirements by setting

$$V_i = \begin{pmatrix}\text{average link i circuit} \\ \text{carried load}\end{pmatrix} + CS_i,$$

for each link i.

3.4.4 Topological Design of Integrated Networks

Use the routing and capacity assignment problem as the inner loop design subproblem, there are two general approaches for topological design of communication networks: the branch exchange method and the concave branch elimination methods. Both are essentially methods for obtaining more cost-effective topologies from a given topology. In the following we describe these two approaches in the context of integrated network design.

3.4.4.1 The Branch Exchange Method

The branch exchange methods was initially proposed in [LIN, 1965] as a heuristic solution method for the traveling salesman problem. Later on, variations of this method have been applied to the design of reliable networks [STEIGLITZ, 1969], natural gas pipeline networks [FRANK, 1969], and the packet switched networks [FRANK, 1970], [GERLA, 1974], [LAVIA, 1975]. This method starts from an arbitrary topological configuration and reaches local minima by means of local transformations. Basically, each iteration of a branch exchange algorithm (in the context of integrated network design) consists of three main steps.

<u>Step 1</u>. Start from a given feasible topology, delete and add one or more links such that the network topology remains feasible (i.e., connectivity constraint, degree constraint, etc. are satisfied).

<u>Step 2</u>. Do routing and capacity assignment on the new topology for the given traffic requirements. If there is a cost improvement, the new topology is accepted. Otherwise, it is rejected.

<u>Step 3</u>. If all feasible local transformations of link exchange have been explored, stop. Otherwise go to Step 1.

The above iterative steps can be repeated with different initial topologies to obtain several local minima; the topology is selected from the local minima.

In [LAVIA, 1975], using graph theoretic techniques, a method has been proposed to find feasible local transformations with only one link addition and/or deletion from a given feasible topology, where feasibility means:

1. The network is *k-connected, for some specific* k,

2. Each pair of nodes is at most d arcs away, for some positive integer d.

These topological constraints are meaningful for most communications networks.

In [GERLA, 1974], a cut-saturation branch exchange method has been proposed. This method tries to identify the branch exchanges that are likely to result in network cost and/or performance improvement, instead of performing exhaustively all possible exchanges. It is similar to [LAVIA, 1975] in the regard that they both try to avoid the unnecessary exchanges, and the two methods can be applied in combination to further reduce the branch exchange candidates.

The cut-saturation method is based on the notion of the saturated cut. The saturated cut is the cut set formed from the most utilized links, and corresponds to the traffic bottleneck in the network. In each iteration of the procedure, either a link (or several links) is added across the cut, or some least utilized links are deleted while maintaining the feasibility of the topology.

3.4.4.2 The Concave Branch Elimination Method

Notice that in the routing and capacity assignment procedures described in Section 3.4.3.4 once a link has been assigned 0 capacity, its capacity will remain 0 throughout the procedure, i.e., the link has been "eliminated." Consequently, an effective topological design method is simply applying the capacity and flow assignment procedure to different initial topologies, and selecting the minimal cost design thus obtained. This approach has been previously applied to communications network design, e.g. [KNEPLEY, 1973], [YAGED, 1971], [GERLA, 1974].

Since the *integrated network* routing and capacity assignment is an involved process (mainly due to the complexity of the circuit routing and flow assignment), the concave branch elimination approach appears to be more effective than the branch exchange approach.

3.5 A BASELINE INTEGRATED NETWORK DESIGN PROGRAM

A program, based on the solution approach described in Section 3.4.3.4, has been developed for the routing and capacity assignment in integrated networks. This program, called INTCAP, can also be used for the topological design of integrated networks. In fact, the program has been extended to handle the routing and capacity assignment of multiple classes of packet requirements, with or without priority. Detailed algorithmic and program description for the multi-class design procedure, however, will be delayed to the next report.

3.5.1 Basic Program Assumptions and Capabilities

The basic assumptions used in the program are:

1. Initial network topology is given.

2. Link costs are given by the power law formula.

3. Packet and protocol overhead (e.g., headers, "RFNM", etc.) are modeled as a constant fraction of the packet traffic requirements.

4. Routing overhead is modeled as a constant fraction of the link capacity.

5. Two options for capacity assignment of signaling messages are available: separate capacity or shared capacity with regular packet traffic.

6. Fixed boundary or movable boundary frame management strategies are available options in the link capacity assignment. These techniques were analyzed in Chapter 2.

7. Packet link delay (for signaling or regular packets) assumes constant blocking probability, constant packet size, random arrivals of signaling and regular packets (see Chapter 2).

Briefly, the program is carried out in the following sequence:

1. Given an initial topology, optimize the cost for the circuit-switched subnet;

2. Calculate the signaling requirements;

3. Optimize the cost for the link signaling capacities;

4. Optimize the total cost for the integrated network.

If the signaling messages share capacity with the regular packet traffic, then Step 3 is merged in Step 4. Steps 1 - 4 can be repeated with different initial optologies to obtain cost improvements.

3.5.2 Designer Options

The following list of user modifiable keywords indicates the options available to the network designers.

Keywords

ITMAX	-	number of routing iterations.
THACC	-	relative throughput accuracy in the packet-switching routing subroutine EXTREM.
H	-	average circuit call holding time (sec).
CONST	-	coefficient in the power law cost formula.
AP	-	exponent in the power law cost formula.
FIXC	-	distance-independent charge in the cost formula.
PRC	-	packet processing delay (sec) at the switch.
PROG	-	packet propagation delay (sec/mile).
PRCLi	-	protocol overhead for packet data class i, i=1,2,...,6.
RTOVR	-	packet routing overhead.
NITRN	-	maximum number of binary search in the capacity assignment subroutine GETCAP to obtain link capacities.
NITRT	-	maximum number of β-T iterations in the capacity assignment subroutine to satisfy the delay constraint.
IBND	-	flag for link channel boundary options (movable or fixed).
VD	-	voice digitation rate (Kbps/voice circuit).
DELT	-	time delay accuracy (sec).
COSTC	-	cost reduction relative accuracy.
DTSi	-	traffic scale factor for packet data class i, i=1,2,...,6.
IPRIO	-	priority flag for signaling and packet data classes (yes or no).
NPC	-	number of packet data priority classes (excluding signaling).

MXSD	-	flag for mixing signaling and packet data traffic (yes or no).
DLYS	-	average signaling set up delay constraint (sec).
DLYi	-	average end-to-end packet delay (sec) constraint for data class i, i=1,2,...,6.
PKLS	-	signaling packet length
PKLi	-	packet length for data class i, i=1,2,...,6.
IDSGN	-	initial packet data flow design options (single, progressive, or min-hop).
KCOMP	-	cost comparison options in the cost reduction loop (class by class or overall classes).

There are two types of keywords. The first type are the design related constraints such as IBND (movable boundary or fixed boundary for the integrated link), DLYi (regular packet average end-to-end delay constraint), H (average circuit call holding time), VD (voice digitation rate). The issues involved in determining the proper values for these keywords are either in the cost-performance trade-offs, (e.g. smaller TMAX implies higher network cost), or in the switching-transmission cost trade-offs (larger values of VD implies higher link bandwidth required, and hence higher transmission cost); however, the switch complexity is reduced, and the cost of devices for generating low voice digitization rates either at the terminal and/or at the switch is also reduced). The second type of keywords are the program accuracy related parameters, such as ITMAX (number of packet routing iterations), DELT (time delay accuracy), THACC, NITRN, etc. The trade-offs involved here are the design accuracy (in relation to the design objectives) versus the program running time. Notice that consistent values should be assigned to the related keywords. For example, it would be irrational in the packet

switching routing to request stringent accuracy constraints but only few routing iterations. The general flow diagram of the main program is shown in Figure 3.

Appendix C presents the program structure, flow diagram, inputs and outputs of the INTCAP program.

3.5.3 An Example

We exhibit movable (keyword IBND = 0) and fixed (IBND = 1) link-boundary network designs for the AUTOVON voice requirement and AUTODIN data requirement. The AUTOVON and AUTODIN system characteristics are as follows:

AUTOVON

Number of user locations	=	665
Number of (lower level) switch locations	=	56
Total traffic	=	2700 Erlangs

AUTODIN

Number of user locations	=	300
Number of switch locations	=	8
Total traffic	=	1436 Kbps

In the present design example, the 8 AUTODIN backbone switches are used as the integrated backbone network switch locations. The AUTOVON requirements on the 8 switches is obtained by homing the 56 AUTOVON switches onto the 8 AUTODIN switches.

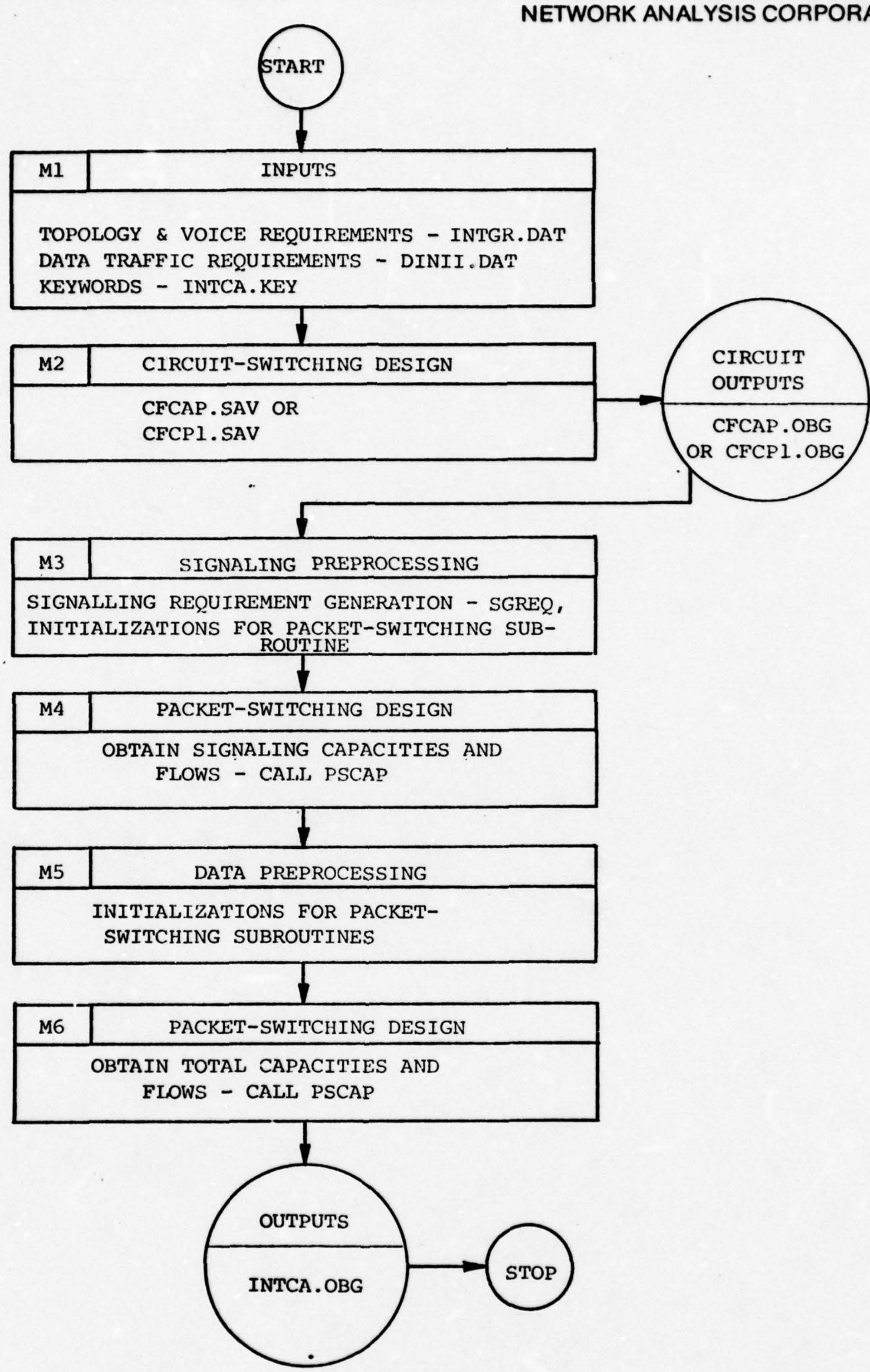


FIGURE 3: MAIN PROGRAM OF INTCAP

The integrated network designs exhibited here were obtained for the case in which only a single class of packet traffic is present.

The following set of keyword values were used:

SMAX (maximum round trip signaling delay)	= .100 (sec)
TMAX (maximum end-to-end data packet delay)	= .100 (sec)
H (average circuit call holding time)	= 30.0 (sec)
PKLHS (average signaling packet length)	= .20 (Kb)
PKLHD (average data packet length)	= .50 (Kb)
CONST (link capacity/cost factor)	= .274
AP (link capacity/cost exponent)	= .50
FIXC (link fixed end cost)	= 417 ($/link)
PRC (packet processing delay)	= .001 (sec)
PROG (line propogation delay)	= .0000008 (sec/mile)
PROVR (data packet protocol overhead)	= .35
RTOVH (data packet routing overhead)	= .07
VD (voice digitation rate)	= 8.0 (Kbps)
DELT (cost difference bound in PSCAP)	= .001
THACC (throughput delay accuracy)	= .001
ITMAX (number of routing iterations in EXTREM)	= 10
NITRN (maximum number of Netwon iterations in DCAP)	= 15
NITRT (maximum number of -T iterations in PSCAP)	= 15
IBND (voice/data boundary policy)	= 0 or 1

The initial input network topology with 8 nodes and 36 links is shown in Figure 4. Input AUTOVON voice requirement (unit Erlang), AUTODIN data requirement (unit packet/sec), and the calculated

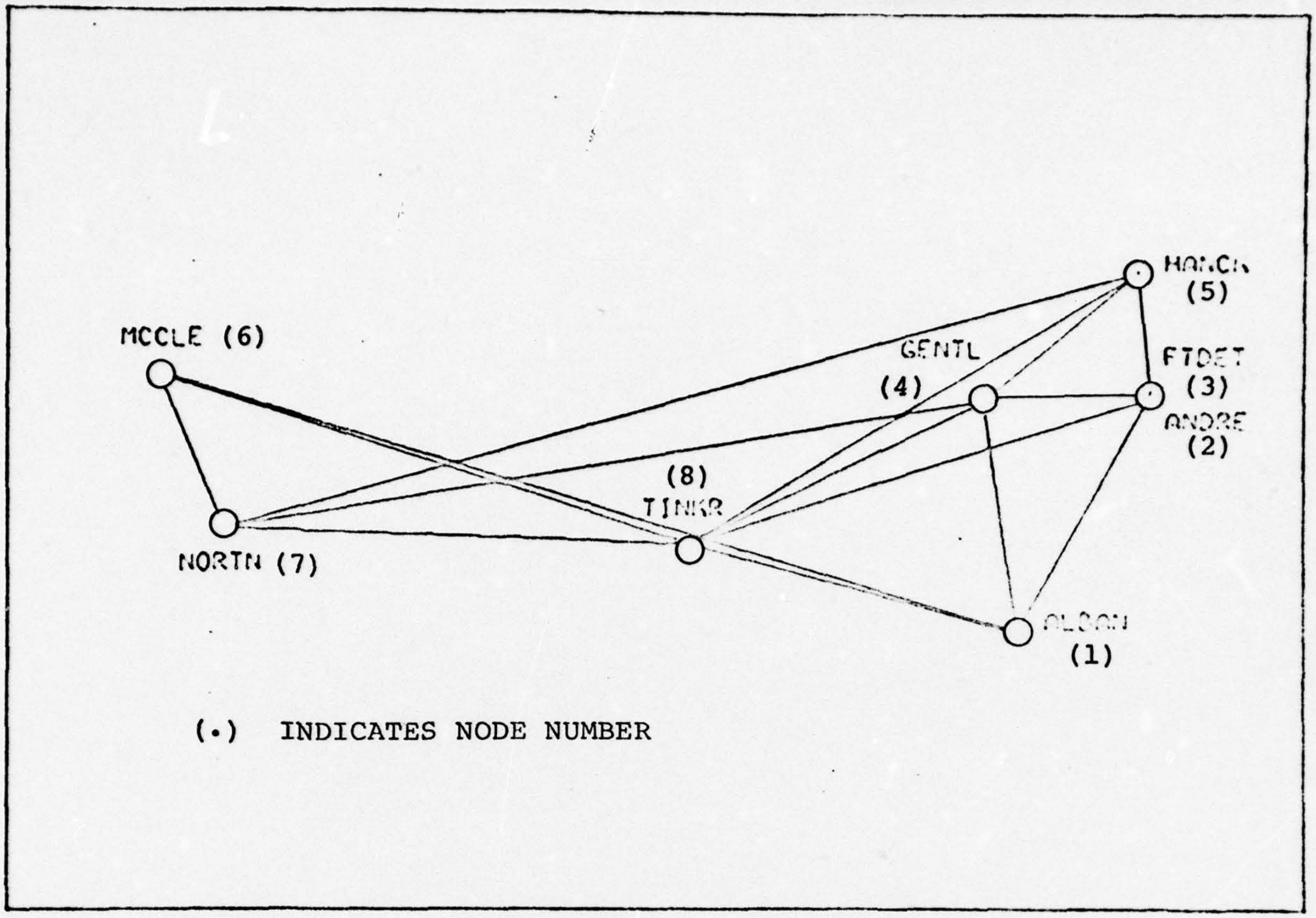


FIGURE 4: SAMPLE DESIGN INITIAL NETWORK TOPOLOGY

signaling requirement (unit packet/sec) are listed in Table 1. The design results for the movable boundary case (IBND = 0) are listed in Table 2. The total link cost is $343.8K/mo. Here the units for flows and capacities are expressed in terms of packets/sec. Several interesting observations can be concluded:

a. There exist links with zero total capacities (links 5 and 27), hence they can be deleted from the input network.

b. There are several links (links 6, 28, 29 and 30) with only packet-switched flow and no circuit-switched flow; but there is no link with only circuit-switched flow and no packet-switched flow.

c. Signaling capacities are small; in fact, they amount to less than 4% of the total capacities.

d. The sum of voice capacity, signaling capacity and data flow for each link is always less than the total capacity, but the difference is small.

For fixed link-boundary case (IBND = 1) a slightly more expensive design is obtained. The results are listed in Table 3. The total link cost is $347.7K/mo. Again observations a. - d. hold for this case. The total capacity of each link is larger than the movable link-boundary case, as expected.

To compare the above two designs, one should take the nodal switch cost into account. In general, movable link-boundary policy requires more powerful switches than fixed link-boundary policy, hence the savings gained in the link costs should be compared against the increased switch costs.

NODE	NODE	DISTANCE	VOICE REQ.	SIG. REQ.	DATA REQ.
1	1	0.000	0.000	0.000	0.000
1	2	644.113	42.000	1.503	38.658
1	3	661.089	24.000	0.160	2.691
1	4	561.722	54.000	1.666	12.856
1	5	908.916	36.000	0.240	11.275
1	6	2152.122	42.000	0.280	7.454
1	7	1926.655	30.000	0.200	15.843
1	8	810.803	72.000	2.379	60.566
2	1	644.113	42.000	1.503	59.943
2	2	0.000	0.000	0.000	0.000
2	3	50.251	28.000	1.297	21.603
2	4	394.207	63.000	2.064	35.055
2	5	300.665	42.000	1.315	59.481
2	6	2379.260	49.000	0.327	28.995
2	7	2252.333	35.000	0.233	28.889
2	8	1153.656	84.000	2.266	53.705
3	1	661.089	24.000	0.160	2.624
3	2	50.251	28.000	1.297	22.908
3	3	0.000	0.000	0.000	0.000
3	4	362.255	36.000	2.144	4.366
3	5	263.091	24.000	0.884	3.496
3	6	2343.926	28.000	0.187	7.592
3	7	2223.631	20.000	0.133	0.200
3	8	1132.862	48.000	0.320	12.024
4	1	561.722	54.000	1.666	21.448
4	2	394.207	63.000	2.064	41.261
4	3	362.255	36.000	2.144	4.805
4	4	0.000	0.000	0.000	0.000
4	5	478.387	54.000	2.551	13.530
4	6	1985.570	63.000	0.420	26.209
4	7	1861.668	45.000	2.046	16.923
4	8	784.357	108.000	5.207	37.337
5	1	908.916	36.000	0.240	11.859
5	2	300.665	42.000	1.315	71.333
5	3	263.091	24.000	0.884	6.065
5	4	478.387	54.000	2.551	13.063
5	5	0.000	0.000	0.000	0.000
5	6	2360.623	42.000	0.280	10.902
5	7	2285.807	30.000	0.200	10.726
5	8	1253.892	72.000	0.480	29.240
6	1	2152.122	42.000	0.280	10.504
6	2	2379.260	49.000	0.327	22.419
6	3	2343.926	28.000	0.187	6.807
6	4	1985.570	63.000	0.420	9.609
6	5	2360.623	42.000	0.280	8.457
6	6	0.000	0.000	0.000	0.000
6	7	392.973	35.000	1.265	14.717
6	8	1341.511	84.000	4.113	35.265
7	1	1926.655	30.000	0.200	21.662
7	2	2252.333	35.000	0.233	36.593
7	3	2223.631	20.000	0.133	0.174
7	4	1861.668	45.000	2.046	10.209
7	5	2285.807	30.000	0.200	16.273
7	6	392.973	35.000	1.265	23.867
7	7	0.000	0.000	0.000	0.000
7	8	1130.533	60.000	1.956	19.586
8	1	810.803	72.000	2.379	76.911
8	2	1153.656	84.000	2.266	73.579
8	3	1132.862	48.000	0.320	12.236
8	4	784.357	108.000	5.207	79.204
8	5	1253.892	72.000	0.480	47.870
8	6	1341.511	84.000	4.113	76.309
8	7	1130.533	60.000	1.956	37.872
8	8	0.000	0.000	0.000	0.000

TABLE 1: TRAFFIC REQUIREMENTS AND LINK LENGTH FOR THE SAMPLE INTEGRATED NETWORK DESIGN

NODE	NODE	LINK NO.	LINK LEN.	VOICE FLOW	VOICE CAP.	SIG. FLOW	SIG.CAP.	DATA FLOW	TOTAL CAP.
1	2	1	644.113	752.744	792.000	1.503	6.444	52.716	875.948
2	1	2	644.113	714.519	752.000	0.000	0.000	62.651	838.564
1	4	3	561.722	751.380	792.000	2.066	8.510	63.648	888.601
4	1	4	561.722	816.024	856.000	3.569	12.828	100.786	1007.241
1	6	5	2152.122	0.000	0.000	0.000	0.000	0.000	0.000
6	1	6	2152.122	0.000	0.000	0.000	0.000	5.154	5.628
1	8	7	810.803	1167.432	1208.000	2.859	9.724	85.506	1362.102
8	1	8	810.803	1111.332	1152.000	2.859	9.724	108.351	1326.350
2	3	9	50.251	678.560	720.000	1.297	9.135	68.971	817.104
3	2	10	50.251	653.400	696.000	6.439	28.986	70.333	812.870
2	4	11	394.207	990.728	1032.000	11.536	34.503	105.328	1220.026
4	2	12	394.207	982.408	1024.000	4.891	17.839	94.113	1180.917
2	5	13	300.665	657.188	696.000	0.000	0.000	56.397	767.691
5	2	14	300.665	584.418	624.000	0.000	0.000	94.022	730.605
2	8	15	1153.656	976.175	1016.000	0.000	0.000	98.083	1158.906
8	2	16	1153.656	1071.591	1112.000	0.000	0.000	106.004	1270.361
3	4	17	362.255	1181.955	1224.000	0.000	0.000	55.061	1334.461
4	3	18	362.255	1102.531	1144.000	0.000	0.000	58.614	1252.680
3	5	19	263.091	395.505	432.000	0.000	0.000	29.057	456.874
5	3	20	263.091	472.920	512.000	5.142	20.214	28.447	561.019
4	5	21	478.387	1270.395	1312.000	8.893	26.968	106.932	1515.098
5	4	22	478.387	1358.746	1400.000	3.751	13.911	70.205	1556.461
4	7	23	1861.668	1016.588	1056.000	2.612	6.919	68.911	1178.287
7	4	24	1861.668	1078.083	1120.000	1.684	4.885	52.439	1224.599
4	8	25	784.357	2700.366	2744.000	9.487	25.082	222.976	3205.253
8	4	26	784.357	2683.907	2728.000	10.415	27.031	309.495	3283.654
5	7	27	2285.807	0.000	0.000	0.000	0.000	0.000	0.000
7	5	28	2285.807	0.000	0.000	0.000	0.000	39.328	42.587
5	8	29	1253.892	0.000	0.000	0.000	0.000	31.767	34.539
8	5	30	1253.892	0.000	0.000	0.000	0.000	2.438	2.689
6	7	31	392.973	612.741	656.000	1.265	6.564	14.035	680.628
7	6	32	392.973	625.426	664.000	4.350	16.341	53.675	748.042
6	8	33	1341.511	2203.027	2248.000	8.691	20.233	137.175	2565.927
8	6	34	1341.511	2060.271	2104.000	5.606	14.153	201.983	2474.671
7	8	35	1130.533	886.160	928.000	0.000	0.000	62.232	1023.048
8	7	36	1130.533	981.541	1024.000	2.156	7.032	120.417	1196.448

TABLE 2: SAMPLE INTEGRATED NETWORK DESIGN RESULT
MOVABLE BOUNDARY STRATEGY

NODE	NODE	LINK NO.	LINK LEN.	VOICE FLOW	VOICE CAP.	SIG. FLOW	SIG.CAP.	DATA FLOW	TOTAL CAP.
1	2	1	644.113	752.744	792.000	1.503	6.444	52.619	918.082
2	1	2	644.113	714.519	752.000	0.000	0.000	63.506	879.796
1	4	3	561.722	751.380	792.000	2.066	8.510	61.804	930.284
4	1	4	561.722	816.024	856.000	3.569	12.828	102.076	1047.571
1	6	5	2152.122	0.000	0.000	0.000	0.000	0.000	0.000
6	1	6	2152.122	0.000	0.000	0.000	0.000	3.629	3.958
1	8	7	810.803	1167.432	1208.000	2.859	9.724	87.190	1407.568
8	1	8	810.803	1111.332	1152.000	2.859	9.724	107.474	1369.125
2	3	9	50.251	678.560	720.000	1.297	9.135	67.308	859.701
3	2	10	50.251	653.400	696.000	6.439	28.986	79.976	869.022
2	4	11	394.207	950.728	1032.000	11.536	34.503	96.814	1255.104
4	2	12	394.207	982.408	1024.000	4.891	17.839	97.087	1228.807
2	5	13	300.665	657.188	696.000	0.000	0.000	57.976	813.572
5	2	14	300.665	584.418	624.000	0.000	0.000	85.936	763.905
2	8	15	1153.656	976.175	1016.000	0.000	0.000	108.766	1213.302
8	2	16	1153.656	1071.591	1112.000	0.000	0.000	104.508	1312.217
3	4	17	362.255	1181.955	1224.000	0.000	0.000	66.039	1391.591
4	3	18	362.255	1102.531	1144.000	0.000	0.000	82.797	1323.494
3	5	19	263.091	395.505	432.000	0.000	0.000	34.724	502.061
5	3	20	263.091	472.920	512.000	5.142	20.214	32.214	606.921
4	5	21	478.387	1270.395	1312.000	8.893	26.968	106.953	1559.819
5	4	22	478.387	1358.746	1400.000	3.751	13.911	82.536	1614.190
4	7	23	1861.668	1016.588	1056.000	2.612	6.919	74.504	1226.782
7	4	24	1861.668	1078.083	1120.000	1.684	4.885	62.591	1280.733
4	8	25	784.357	2700.366	2744.000	9.487	25.082	202.064	3229.837
8	4	26	784.357	2683.907	2728.000	10.415	27.031	299.544	3320.632
5	7	27	2285.807	0.000	0.000	0.000	0.000	0.000	0.000
7	5	28	2285.807	0.000	0.000	0.000	0.000	26.260	28.422
5	8	29	1253.892	0.000	0.000	0.000	0.000	17.267	18.777
8	5	30	1253.892	0.000	0.000	0.000	0.000	1.751	1.928
6	7	31	392.973	612.741	656.000	1.265	6.564	10.414	723.074
7	6	32	392.973	625.426	664.000	4.350	16.341	40.523	774.931
6	8	33	1341.511	2203.027	2248.000	8.691	20.233	140.115	2617.967
8	6	34	1341.511	2060.271	2104.000	5.606	14.153	212.930	2534.049
7	8	35	1130.533	886.160	928.000	0.000	0.000	56.430	1061.821
8	7	36	1130.533	981.541	1024.000	2.156	7.032	96.573	1216.333

TABLE 3: SAMPLE INTEGRATED NETWORK DESIGN RESULT
FIXED BOUNDARY STRATEGY

We also obtained a network design using only min-hop routes for the packet switching design, and found the resulting cost for the packet capacities is at least 50% more expensive than the two cases presented here.

3.5.4 Future Program Development

The capabilities of the program currently implemented for integrated network design were presented in previous sections. The structure of the program, its flow diagrams, inputs, outputs, and subroutines are detailed in Appendix C. Thus far the program was applied to design relatively small networks with the objectives of improving computational efficiency and verifying convergence. Future design experiments will be directed at optimum network designs and will guide program restructuring and modifications to obtain minimum cost network design at maximum computational efficiency. Apart from program developments that will be suggested by further experimentation, several specific extensions that will be implemented are now outlined:

1. Incorporation of an integrated node model in the integrated design program. This will enable the determination of the circuit and packet switching capacities, and consequently the costs, of switching nodes.

2. Implement discrete cost functions for link and switch cost models; either directly in the design or via the analysis loop (outer loop), by transforming continuous capacities to discrete values using some criterion. At present, concave cost models of cost as a function of capacity are implemented.

3. Changing the general design procedure to one in which the design is done in parallel for the circuit switched and packet switched traffic. At present it is performed sequentially, first the circuit switched subnet and next the packet switched subnet.

REFERENCES

[CANTOR, 1974] Cantor, D.G., and M. Gerla, "Optimal Routing in a Packet-Switched Computer Network," IEEE Transactions on Computers, Vol. C-23 (10), October 1974, pp. 1062-1069.

[COVIELLO, 1975] Coviello, G., and P. Vena, "Integration of Circuit/Packet Switching by a SENET (Slotted Enveloped Network) Concept," Proceedings of the National Telecommunications Conference, December 1975.

[GERLA, 1975] Gerla, M., H. Frank, W. Chou and J. Eckl, "Design Alternatives for Large Distributed Networks," Proceedings of the National Telecommunications Conference, New Orleans, Louisiana, December 1975.

[KATZ, 1967] Katz, S.S., "Statistical Performance Analysis of a Switched Communication Network," Fifth International Teletraffic Congress, New York, June 1967.

[KLEINROCK, 1972] Kleinrock, L., Communication Nets: Stochastic Message Flow and Delay, Dover, New York, 1972.

[KNEPLEY, 1973] Knepley, J.E., "Minimum Cost Design for Circuit Switched Networks," DCA Technical Note No. 36-73, July 1973.

REFERENCES (Cont'd)

[MIYAHARA, 1975] Miyahara, H., T. Hasegawa, and Y. Teshigawara, "A Comparative Evaluation of Switching Methods in Computer Communication Networks," Proceedings of International Conference on Communications, 1975.

[NAC, 1976a] Network Analysis Corporation, Sixth Semiannual Technical Report, Vol. 3, Chapter 2, January 1976.

[FISHER, 1976] Fisher, M.J. and T.C. Harris, "A Model for Evaluating the Performance of an Integrated Circuit and Packet Switched Multiplex Structure," IEEE Trans. on Communications, February 1976, pp. 195-202.

[KARP, 1975] Karp, R.M., "On the Computational Complexity of Combinatorial Problems," Networks, 5, (1975), pp. 45-68.

[NAC, 1976b] Network Analysis Corporation, Seventh Semiannual Technical Report, August 1976.

[FRATTA, 1973] Fratta, L., M. Gerla, and L. Kleinrock, "The Flow Deviation Method: An Approach to Store-and-Forward Communication Network Design," Networks, (3), 1973, pp. 97-133.

REFERENCES (Cont'd)

[HAMMING, 1962] Hamming, R.W., Numerical Methods for Scientists and Engineers, McGraw-Hill, New York, 1962.

[LIN, 1965] Lin, S., "Computer Solutions of the Traveling Salesman Problem," Bell System Tech Journal Vol. 44 (10) 1965, pp. 2245-2269.

[STEIGLITZ, 1969] Steiglitz, G., P. Weiner and D. Kleitman, "Design of Minimum Cost Survivable Networks," IEEE Transactions on Circuit Theory, 1970 Vol. CT-16, pp. 455-460.

[FRANK, 1969] Frank, H. I.T. Frisch and W. Chou, "Topological Considerations in the Design of the ARPA Computer Network," AFIPS National Computer Conference Proceedings, 1970, pp. 581-587.

[GERLA, 1974] Gerla, M., H. Frank, W. Chou and J. Eckl, "A Cut Saturation Algorithm for Topological Design of Packet Switched Communication Networks," Proceedings of National Telecommunications Conference, 1974.

[LAVIA, 1975] Lavia, A., and E.G. Manning, "Perturbation Techniques for Topological Optimization of Computer Networks," Proceedings of the 4th Data Comm. Symp., 1975.

[YAGED, 1971] Yaged, B., Jr., "Minimum Cost Routing for Static Network Models," Networks, (1) 1971, pp 139-172.

APPENDIX A

DERIVATION OF END-TO-END LOSS PROBABILITY AS A FUNCTION OF AVERAGE LINK BLOCKING PROBABILITY

In this appendix we derive the approximate formula (4) relating end-to-end loss to the average link blocking probability.

The average link blocking probability P_{AL} for a network is defined by (see Section 3.3.1)

$$P_{AL} = \frac{1}{NA} \sum_{\ell=1}^{NA} P_{\ell}, \tag{A.1}$$

where

NA = total number of links in the network,

P_{ℓ} = average blocking probability of link ℓ.

We assume the following progressive routing scheme: For each link ℓ on the primary path for a requirement pair (s,t), there is one alternate path from the initial node of ℓ to t which does not contain link ℓ. We call such an alternate path the alternate path branching from link ℓ, and denote it by $R_{s,t}(\ell)$. (see Figure A.1) Moreover, we assume that;

1. The system is in statistical equilibrium.

2. The occupancy distribution of the trunk groups throughout the network are statistically independent of each other.

3. No congestion is encountered at the switching node.

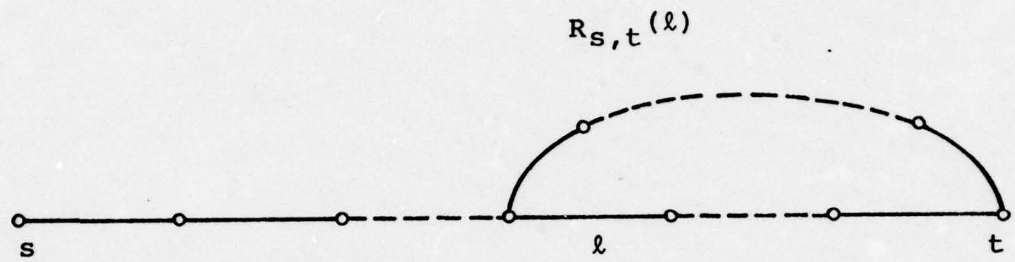


FIGURE A.1: A PROGRESSIVE ROUTING SCHEME

4. All flow groups to a link encounter the same blocking probability.

5. The average blocking probability for a link is small (say $\leq$ 5%).

6. The average number of links on an end-to-end path is small (say, $\leq$ 5).

Consider a requirement pair (s,t). For convenience, let;

$\ell_i^0(s,t)$ = the i^{th} link on the primary path for (s,t);

$R_i(s,t)$ = the alternate path for (s,t) branching from $\ell_i^0(s,t)$;

$\ell_j^i(s,t)$ = the j^{th} link on the path $R_i(s,t)$, $i \geq 1$;

$p_j^i(s,t)$ = the blocking probability of link $\ell_j^i(s,t)$;

$PP_i(s,t)$ = the path blocking probability of $R_i(s,t)$;

$k(0,s,t)$ = number of links on the primary path for (s,t);

$k(i,s,t)$ = number of links on $R_i(s,t)$.

We shall omit the (s,t)'s when there is no danger of confusion. Thus, p_j^i shall stand for $p_j^i(s,t)$.

From Assumptions 1-4, the end-to-end loss probability, $P_E(s,t)$, for requirement (s,t) is given by;

$$P_E(s,t) \approx p_1^0 \times PP_1 + (1-p_1^0)p_2^0 PP_2 + \ldots + (1-p_1^0)(1-p_2^0)\ldots(1-p_{k(0)-1}^0)p_{k(0)}^0 PP_{k(0)}. \tag{A.2}$$

and

$$PP_i \approx 1-(1-p_1^i)(1-p_2^i)\ldots(1-p_{k(i)}^i), \quad i=1,\ldots,k(0) \tag{A.3}$$

By Assumptions 5 and 6, Eqs. (A.2) and (A.3) reduce to;

$$P_E(s,t) \approx \sum_{i=1}^{k(0)} p_i^0 \times PP_i \tag{A.4}$$

$$PP_i \approx p_1^i+p_2^i+\ldots+p_{k(i)}^i, \quad i=1,\ldots,k(0). \tag{A.5}$$

Approximate the p_j^i's by P_{AL}, we obtain

$$P_E(s,t) \approx \sum_{i=0}^{k(0)} \sum_{j=0}^{k(0)} P_{AL}{}^2 \tag{A.6}$$

Let

NL_P = average number of links in a primary path,

NL_A = average number of links in an alternate path.

From equation (A.6), we obtain the desired result:

$$P_E \approx NL_P \times NL_A \times P_{AL}^2 \qquad \text{(A.7)}$$

Equation (A.7) is quoted in Section 3.3.1

APPENDIX B

DERIVATION OF THE AVERAGE PATH BLOCKING PROBABILITY

In this appendix we derive the average path blocking probability - Equation (3), quoted in Section 3.2.2.

For each requirement pair (s,t), let

$\Phi(s,t)$ = the set of end-to-end path from s to t.

Also, for $R \varepsilon \Phi(s,t)$, let

$f_R(s,t)$ = the carried load on R for requirement pair (s,t),

P_R = the blocking probability of path R.

Moreover, let

γ = total carried loads in the network

$$= \sum_{s,t} \sum_{R \varepsilon \Phi(s,t)} f_R(s,t). \qquad \text{(B.1)}$$

We define the average end-to-end path blocking probability P_{AR} by,

$$P_{AR} = \frac{1}{\gamma} \sum_{s,t} \sum_{R \varepsilon \Phi(s,t)} f_R(s,t) P_R \qquad \text{(B.2)}$$

We derive an alternative expression for P_{AR} in terms of the links (which in general reduces the complexity of computation). Due to the complex interdependence of the links in circuit routing, an exact expression will necessarily be very complicated. Here instead an approximate expression is derived using Assumptions 1-6 of Section 3.3.1.

Consider an end-to-end path R consisting of m links, each with blocking probability p_i. Applying Assumptions 1-6 of Section 3.3.1, we can show that:

$$P_R \approx p_i + \dots + p_m. \qquad (B.3)$$

Consequently, if for each link ℓ, we let

p_ℓ = average link blocking probability of ℓ,

f_ℓ = total carried load on link ℓ,

then

$$P_{AR} = \frac{1}{\gamma} \sum_{s,t} \sum_{R\varepsilon\Phi(s,t)} f_R(s,t)\, P_R$$

$$\approx \frac{1}{\gamma} \sum_{s,t} \sum_{R\varepsilon\Phi(s,t)} f_R(s,t) \sum_{\ell\varepsilon R} p_\ell$$

$$= \frac{1}{\gamma} \sum_{\ell=1}^{NA} \sum_{s,t} \sum_{\ell\varepsilon R} f_R(s,t) p_\ell$$

$$= \frac{1}{\gamma} \sum_{\ell=1}^{NA} f_\ell\, p_\ell. \qquad (B.4)$$

We thus obtain (notice that this detivation is similar to Kleinrock's development of the average end-to-end delay formula [KLEINROCK, 1972])

$$P_{AR} \approx \frac{1}{\gamma} \sum_{\ell=1}^{NA} f_\ell p_\ell \qquad (B.5)$$

APPENDIX C

A PROGRAM FOR INTEGRATED PACKET/CIRCUIT SWITCHED NETWORK DESIGN

Briefly, the program is carried out in the following sequence:

1. Optimize the cost for the circuit switched subnet;

2. Calculate the signaling requirements,

3. Optimize the cost for the link signaling capacities; and

4. Optimize the total cost for the integrated network.

C.1 Program Parameters

In the following descriptions of program inputs and outputs, the dimensions of arrays and matrices are explicitly indicated in the parentheses.

INPUTS

Topology ...	MXN	-	maximum number of nodes
	MXL	-	maximum number of links
	NN	-	number of nodes
	NI(NL), NJ(NL)	-	start, end nodes of links
	DIST(NN,NN)	-	distance matrix (mile)
	DI(NL)	-	lengths of links (mile)

Voice requirements ...

EEPB	-	average end-to-end blocking probability constraint
VTRAF(NN,NN)	-	voice traffic requirement matrix (Erlangs)

Data traffic requirement ...

DTR(NN,NN)	-	data traffic requirement matrix (Kbits/sec)

Keywords ...

SMAX	-	average signaling set up delay constraint (sec)
TMAX	-	average end-to-end packet delay constraint (sec)
H	-	average circuit call holding time (sec)
PKLHS	-	signaling packet length
PKLHD	-	data packet length
CONST	-	coefficient in the power law cost formula
AP	-	exponent in the power law cost formula
FIXC	-	distance - independent charge in the cost formula
PRC	-	processing delay/packet at the switch (sec)
PROG	-	propagation delay/packet/unit length (sec/mi)
PROVR	-	average packet protocol overhead
RTOVH	-	packet routing overhead
VD	-	voice digitation rate (bits/circuit)
DELT	-	time delay accuracy (sec)
THACC	-	relative throughput accuracy in the packet-switch routing subroutine EXTERM
ITMAX	-	number of routing iterations
NITRN	-	maximum number of Newton's iterations in the capacity assignment subroutine to obtain the capacities
NITRT	-	maximum number of β - T iterations in the capacity assignment subroutine to satisfy the delay constraint

IBND	-	flag for link channel boundary model
		= 0 movable boundary
		= 1 fixed boundary

All inputs except MXN, MXL, are under user's control.

OUTPUTS

Circuit design ...

PL(NL)	-	link blocking probabilities
NCH(NL)	-	trunk sizes
VCRLK(NL)	-	voice offered loads
COST	-	total link circuit capacity cost

Signaling design ...

SFLK(NL)	-	signaling flows
SCAP(NL)	-	signaling capacity assignments
TC	-	total link signaling capacity cost

Integrated design ...

DFLK(NL)	-	data flows
CCAL(NL)	-	total capacity assignments
TCOST	-	total integrated link capacity cost

The main program of INTCAP interfaces with the circuit network design package by virtue of I/O files only. This is due to the possible size limitation on the available core.

C.2 Program Block Diagrams

C.2.1 Flow Diagrams

In this subsection we exhibit the flow charts for the main, the circuit-switching, and the packet-switching modules (Figures C.1, C.2A, C.2B, C.3). There are two versions of the circuit switching network design package (Figure C.2A, Figure C.2B). Most boxes in the flow charts will be explained in Section C.3.2. The boxes are numbered at the upper left corners, followed by brief functional descriptions. Figure C.4 summarizes the modular structures of all subroutines.

C.3 Subroutine Descriptions

In this subsection, we briefly describe the inputs/outputs, the function and the algorithm for all the subroutines depicted in Figure C.4, except the main routine INTCAP, the subroutine EXTREM and the five I/O files INTCA.KEY, INTGR.DAT, DINII.DAT, DINII.IBG, and INTCA.OBG, which have already been described in Sections C.1 and C.2. For clarity, the subroutines will be classified into four groups, called Circuit Design I, Circuit Design II, Signaling Design and Packet Design. The circuit switch related subroutines were discussed in [NAC, 1976b]. They are presented here for the purpose of self-containment.

Circuit Design I

a. SDROUT

Inputs:	NN
	MARCOS - direct costs
Outputs:	LSTCOS - path costs
	ROUT - primary routes
Function:	Find the shortest paths between all pairs of nodes.
Algorithm:	Floyd shortest path algorithm.

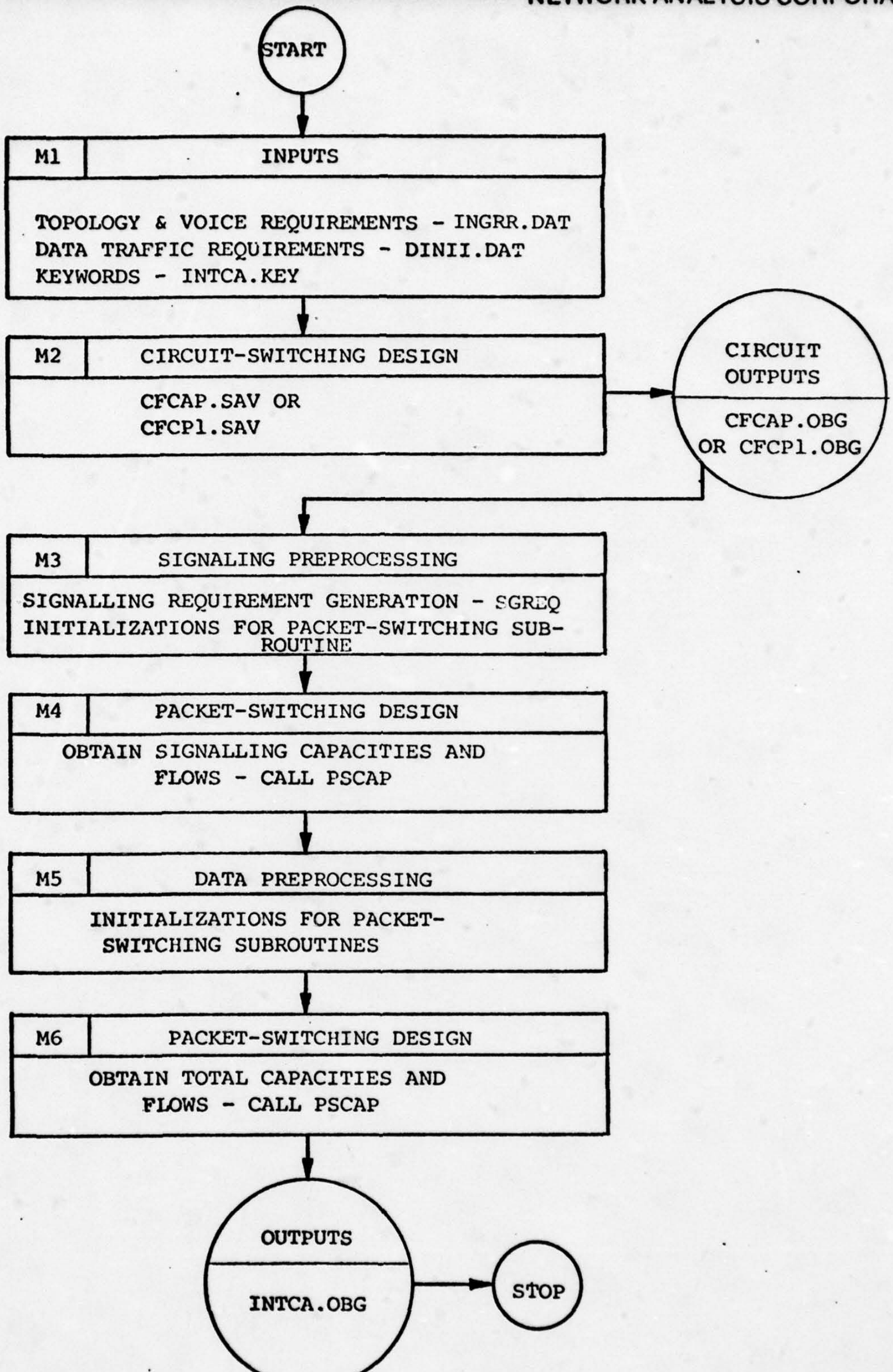


FIGURE C.1: MAIN PROGRAM OF INTCAP

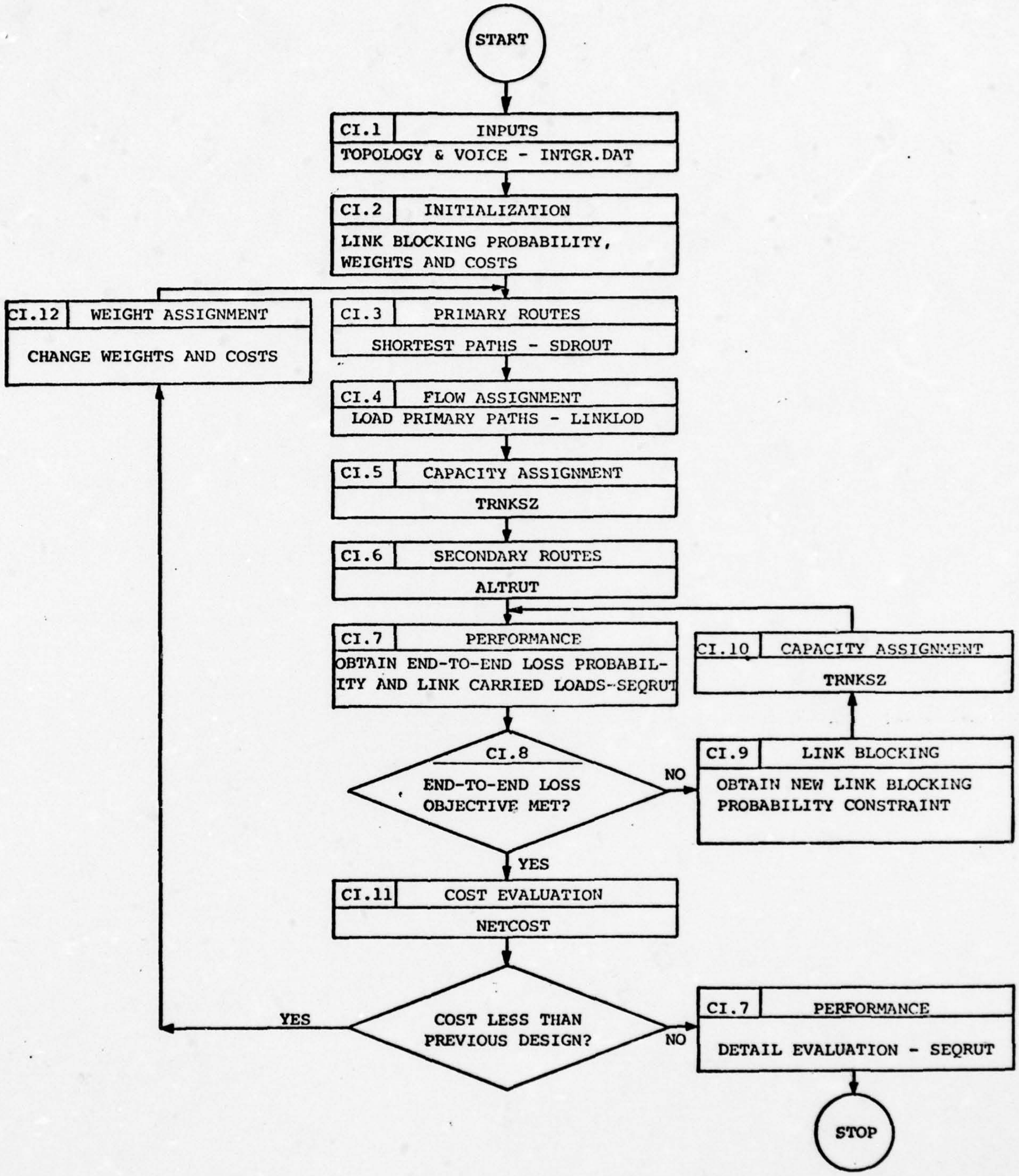


FIGURE C.2A: CIRCUIT-SWITCHING DESIGN (CFCAP.SAV)

FIGURE C.2A: CIRCUIT-SWITCHING DESIGN (CFCAP.SAV)

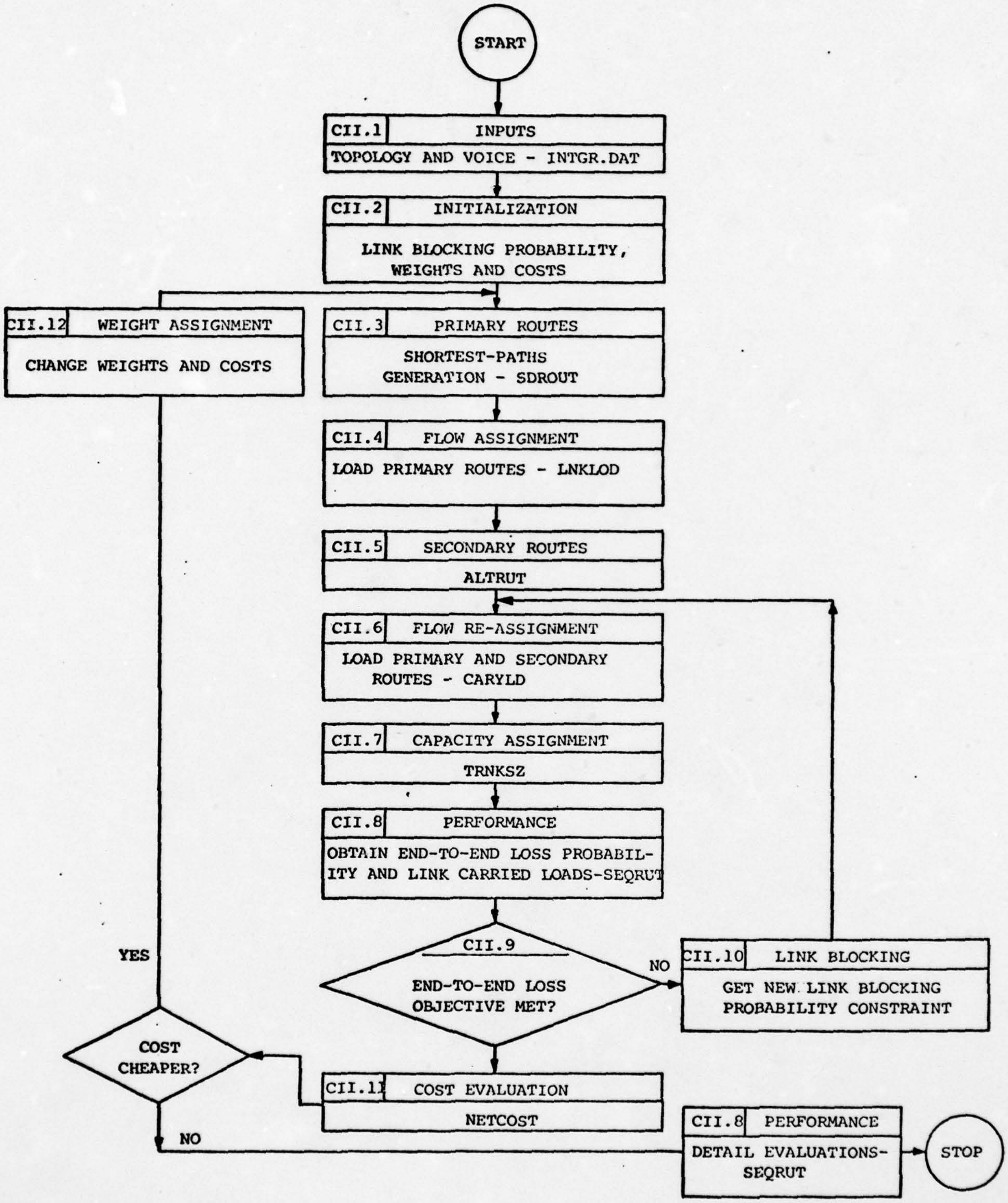


FIGURE C.2B: CIRCUIT-SWITCHING DESIGN (CFCP1.SAV)

FIGURE C.2B: CIRCUIT-SWITCHING DESIGN (CFCP1.SAV)

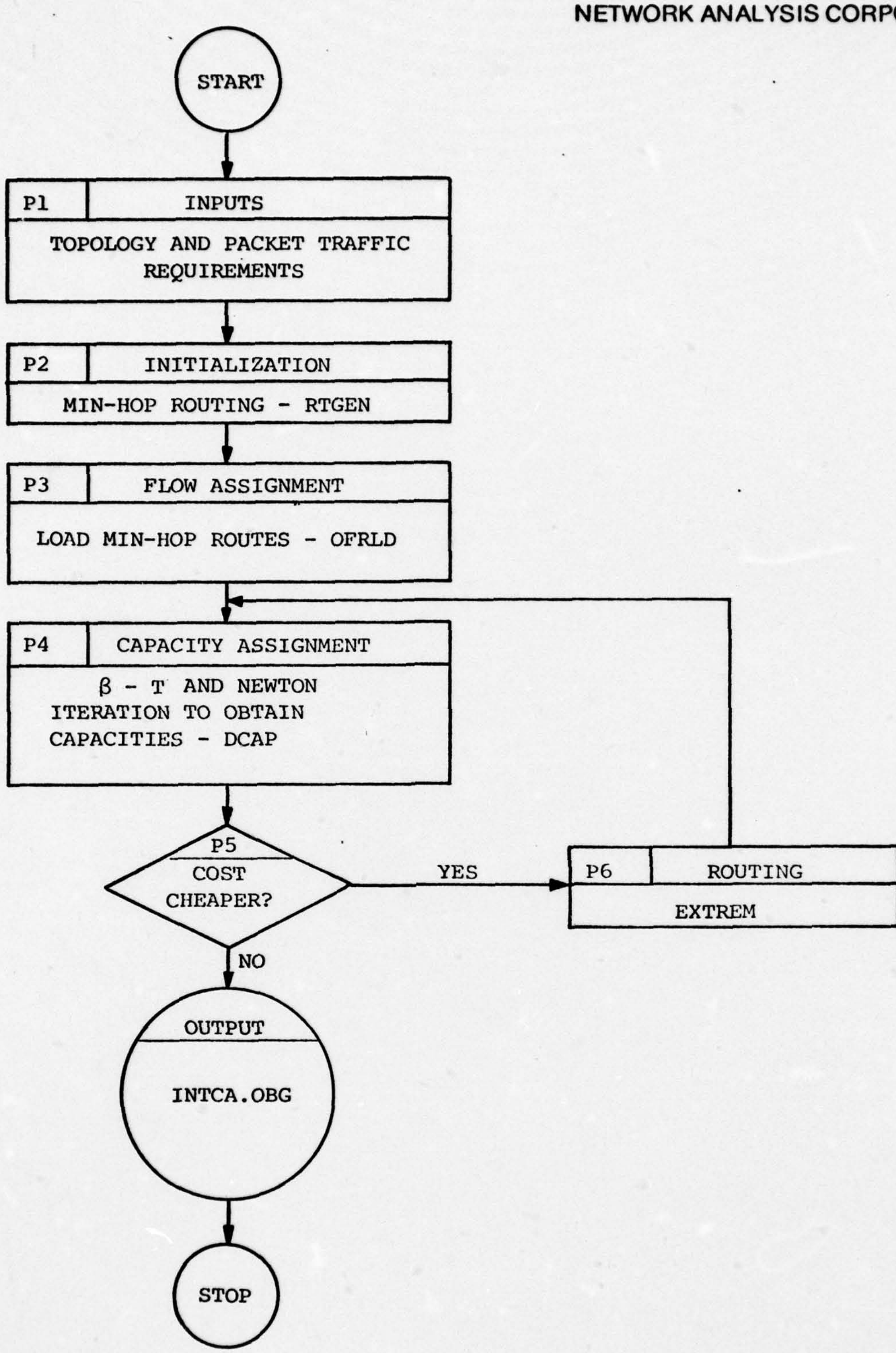


FIGURE C.3: PACKET-SWITCHING DESIGN (PSCAP)

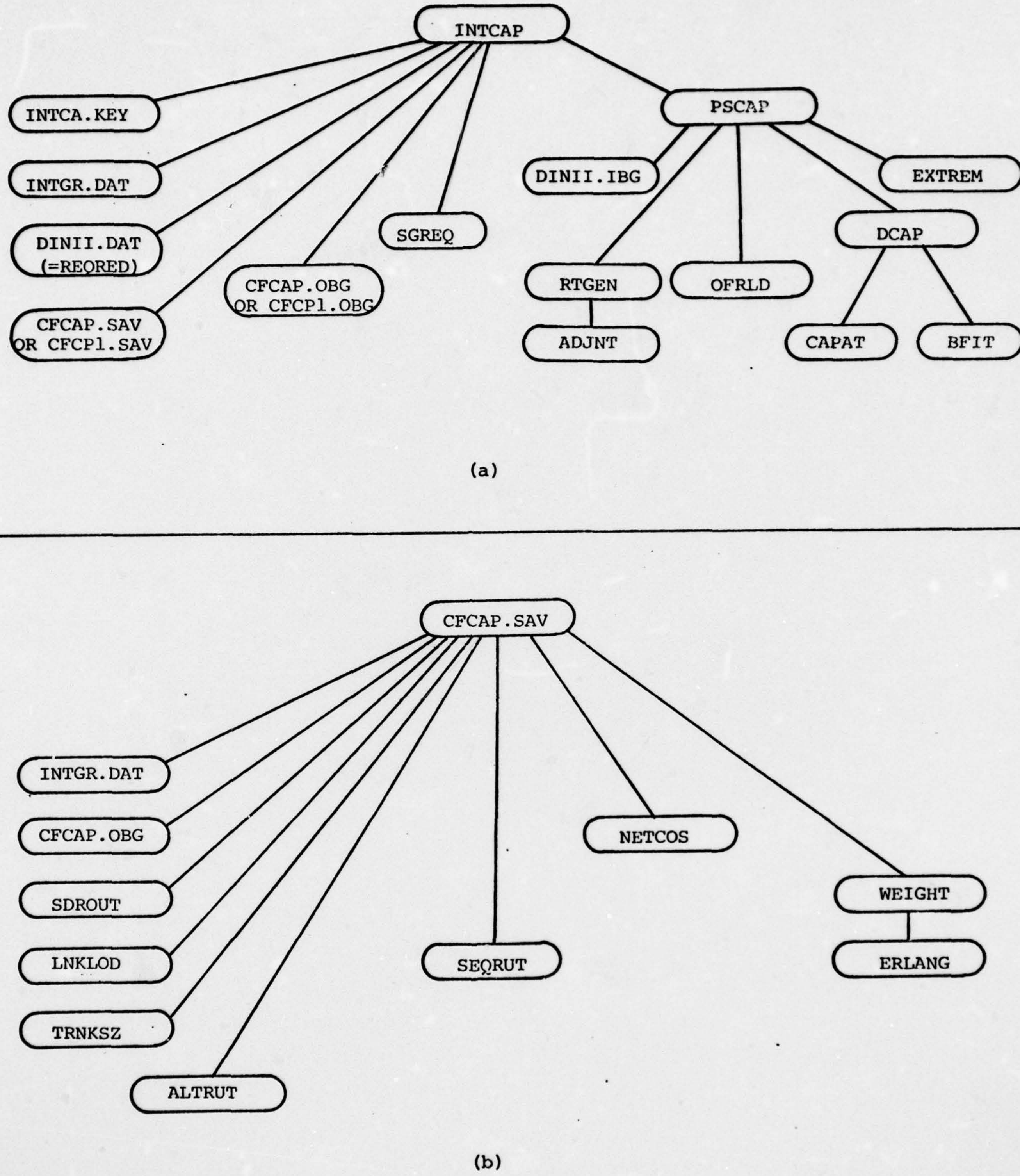
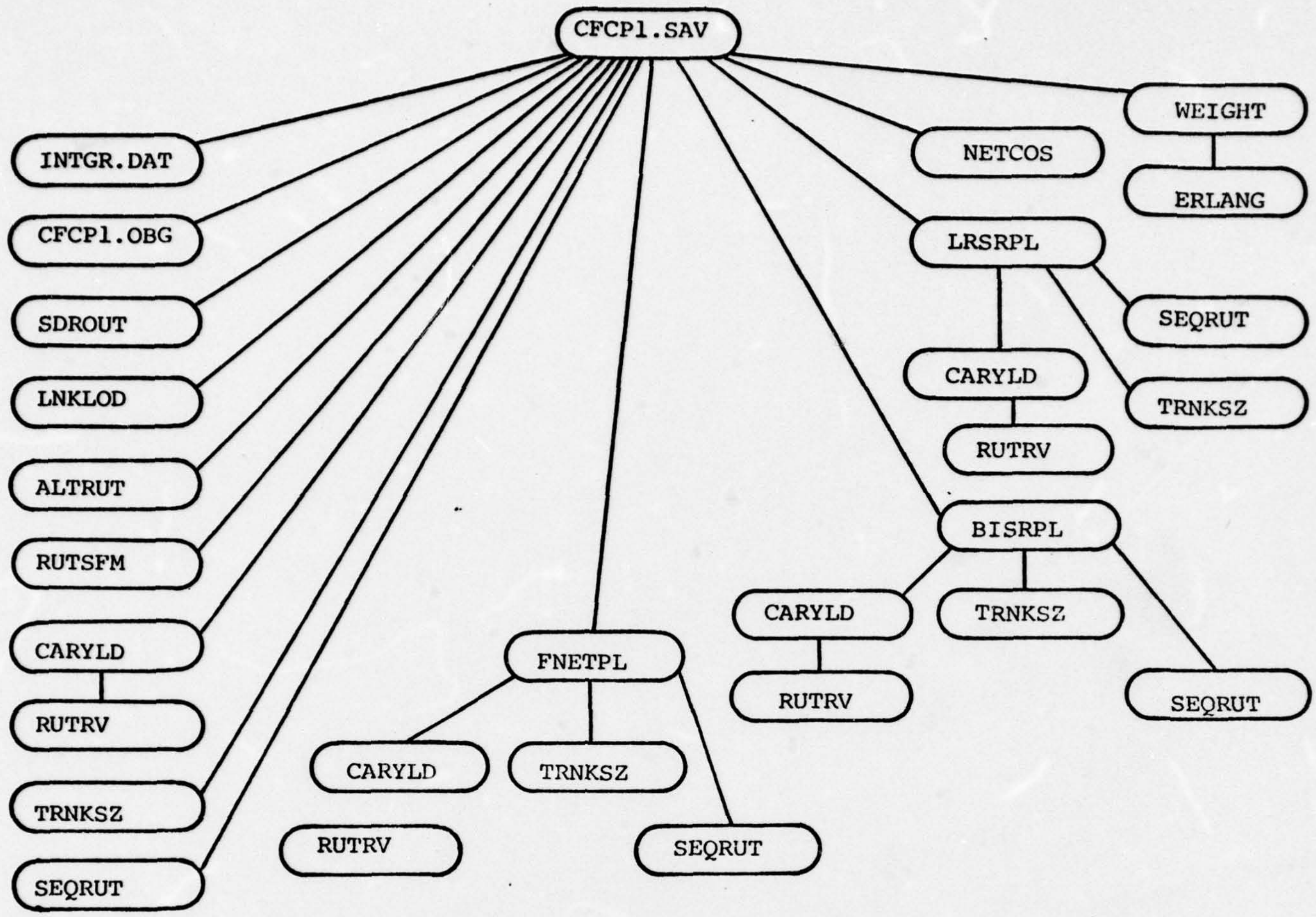
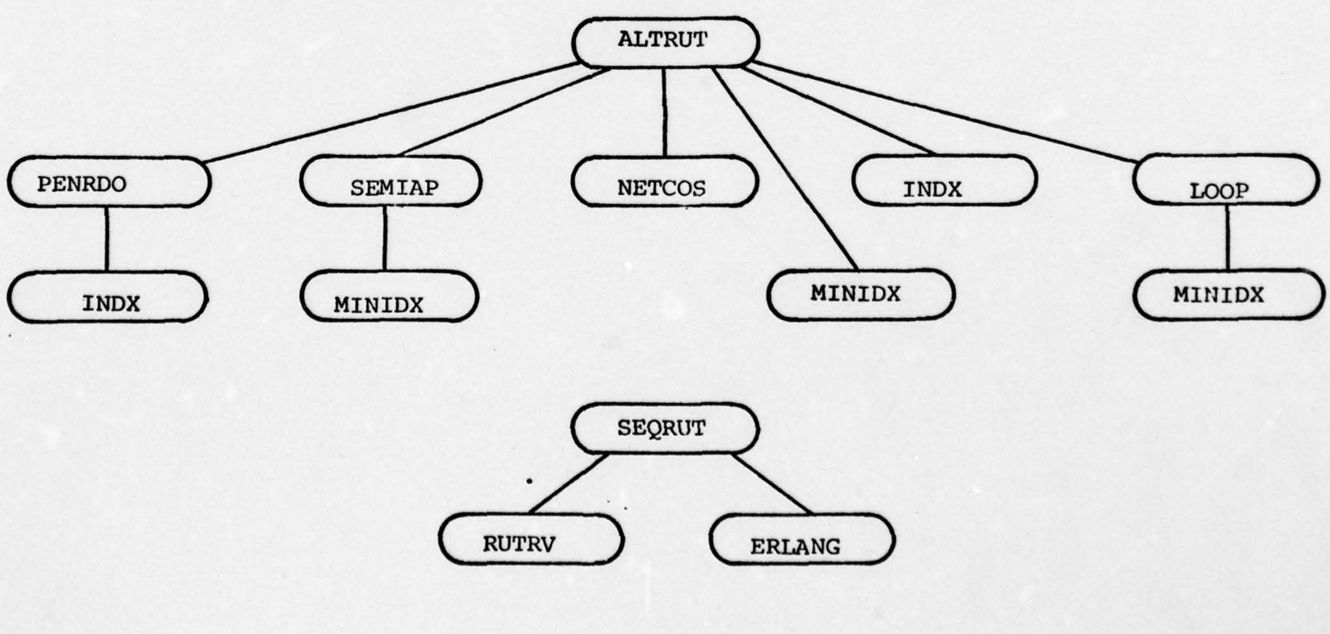


FIGURE C.4: MODULAR STRUCUTRES OF SUBROUTINES



(c)



(d)

FIGURE C.4 (Cont'd)

b. LNKLOD

Inputs:	NN
	VTRAF - Voice traffic requirements
	ROUT - primary routes
Outputs:	LOAD - flows on primary paths
Function:	Assign flows to links on primary paths, based on traffic requirement.
Algorithm:	straight-forward loading of the requirements onto each link of each path.

c. TRNKSZ

Inputs:	NN
	LOAD - primary flows
	F - link blocking probability
Output:	TRUNK - trunk sizes
Function:	Assign link capacitites from the given flows and link blocking probability
Algorithm:	Erlang B loss formula

d. ALTENT

Inputs:	NN, DIST
	R - number of alternate routes
	LSTCOS - pair costs
	TRUNK - trunk sizes
	ROUT - primary paths
Outputs:	ALTVTR - alternate routing vector
	HLTIDX - positions of starting intermediate nodes in ALTVRT
	MN - length of ALTVTR
Function:	Get next best alternate routes for each node pair.

Algorithm: Step 1 From source node fans out to the nearest neighbor not on the primary path,

Step 2 If the nearest neighbor does not exist, insert a link to the nearest neighbor and assign trunk to the link;

Step 3 The alternate path is set to be the union of the link to the nearest neighbor and the primary path from the nearest neighbor to the destination node.

e. SEQRUT

Inputs: NN, BTRAF, ALTIDX, ALTVTR, MN, ROUT, TRUNK

Outputs: AVRBPB - average end-to-end loss probability
SUMA - link offered loads
SUMB - link blocking probabilities

Function: From primary routes and alternate routes, calculate the average end-to-end loss probabilities.

Algorithm: Step 1 Load traffic requirements onto primary and alternate routes.

Step 2 Sum the lost flows on the alternate paths.

Step 3 Divide of the sum by total traffic

Step 4 If the stopping condition is reached, exit; otherwise got to Step 1.

f. NETCOS

Inputs: NN, DIST, TRUNK

Output: COST - sum of link costs

Function: Get network cost by using power law link cost formula.

g. WEIGHT

Inputs: NN, TRUNK, LOAD
Output: W - weights of links
Function: Get new link weights.

h. PENADO

Function: Identify the nodes having only one incoming (or outgoing) link. The purpose is to assign trunks (or flow on the link) connecting this node to its secondary neighbor so as to guarantee alternate paths.

i. SEMIAD

Function: Identify the nodes having one incident (ongoing) link and more than one outgoing (incoming) links. The purpose is the same as h.

j. LOOP

Function: Insure alternate routes not containing the sending nodes.

k. INDX

Function: Get index of the searched element in an array.

l. MINIDX

Function: Get the index of the minimum element in an array.

m. RUTKV

Function: Get all nodes in a path and the length of the path.

n. ERLANG

Function: Get Erlang value of the flow.

Circuit Design II

a-c, f-n. Are the same as Circuit Design I.

d. ALTRUT

Same as I.d, except TRUNK is replaced by LOAD.

o. RUTSFM

Inputs: NN, ROUT, ALTIDX, ALTVTR, MN
Outputs: PRMIDX, PRMVTR, IX, RUTIDX, RUTVTR, IY
Function: Transform primary routes from table form to array form, in order to be consistent with the route data structure of the alternate routes.

e. SEQRUT

Same as I.e, except primary routes input in array form, and LOAD contains contributions from both primary routes and alternate routes.

p. CARYLD

Inputs: NN, VTRAF, PRMIDX, PRMVTR, IX, RUTIDX, RUTVTR, IY

Outputs: SUMA, LOAD - sum of flow on links from primary and alternate routes.

Function: Load links with flows from both primary and alternate routes.

q. FNETPL

Function: Update link blocking probability E using the formula

$$E = E + (\sqrt{EEBP} - \sqrt{AVRBPB}) \times \frac{\text{const.}}{\text{NITR}}$$

where;

EEBP = the required end-to-end blocking,

AVRPBP = the calculated end-to-end loss,

const. = input specified constant

NITR = number of iterations.

r. BISRPL

Function: Get new link blocking probability by the binary search method.

Algorithm: Step 1 Get two values of link blocking probabilities whose corresponding AVRBPB's bound EEPB in between;

<u>Step 2</u> Take the median of the above two E's and calculate its AVRBPB;

<u>Step 3</u> Select two of the above three points which give a tighter bound of EEPB.

<u>Step 4</u> If desired closeness to EEPB has not been reached, go to Step 2, otherwise, end.

s. <u>LRSRPL</u>

Function: Get net link blocking probability by linear fit to AVRBPB - E curve.

Algorithm: Same as the binary search, except in Step 2, the linear fit of AVRBPB - E curve is used. The intersection of EEPB and the linear fit determines the new value of link blocking.

<u>Signaling Design</u>

a. <u>SGREQ</u>

Inputs: NN, NL, NI, NJ, BTRAF, H, PKLHS,
VCRLK - voice carried loads

Outputs: S - signaling traffic requirement
TCALL - total number of calls

Function: From voice traffic requirement and carried loads, get signaling traffic requirement matrix, and total number of calls.

Algorithm: Use Equation (4) of Section 3.4.

Packet Design

a. RTGEN

Inputs: NN, NL, NI, NJ
TRAF - traffic requirement

Outputs: TABLE - min-hop routes
DIRECT - positions of links in min-hop routes.

Function: Get min-hop routes for initial loading of packet traffic.

Algorithm: For each node, perform a breath-first search to get the min-hop tree.

b. OFRLD

Inputs: NN, NL, TABLE, DIRECT, TRAF

Outputs: DFLK - packet flows on links

Function: load traffic requirement on min-hop routes for initial capacity assignment.

Algorithm: Traverse each rooted tree once using the leaf truncation scheme, and load the tree with the requirements.

c. ADJNT

Inputs: NN, NL, NI, NJ

Outputs: IR - node pointers point to the node of the first link of incident links
IK - link pointers point to the last link of incident links

Function: Get adjacent nodes for breath-first search in RTGEN.

d. DCAP

Inputs: NL, TMAX, PKLHD, AP, CONST, FIXC, PAC, PROG, NITRN, NITRT, DELT, PROVR, RTOVH, DFLK
DI - lengths of links
V, Z - occupied flows, capacities
DRE - total traffic requirement

Outputs: CCAL - capacity assignment
TCOST - sum of link costs

Function: From flows and end-to-end delay requirement get capacities and cost.

Algorithm: (see Section 3.4.2)

Step 1 Get two β's whose corresponding delays bound TMAX;

Step 2 Select a third β in between and do either binary search or quadratic fit to get a new β, which gives a new T closer to the TMAX.

Step 3 Selection two β's out of the three which give a tighter bound on TMAX.

Step 4 If the desired closeness to TMAX or the number of iterations has not been reached, go to Step 2; otherwise end.

e. CAPAT

Inputs: NL, DI, TMAX, PKLHD, AP, FIXC, PRC, PROG, V, DFLK, DRE, PROVR, RTOVH, DELT, NITRN, BETA

Outputs:	CCAL
	WAIT - calculated delay for given β.
Function:	Obtain capacities and delay for given input β.
Algorithm:	Given β, solve a system of NA+1 equations to obtain the corresponding average end-to-end delay using Newton's method. (see Section 3.4.2)

f. <u>BFIT</u>

Inputs:	Three sets of β-T values
Outputs:	Coefficients of the quadratic equation
Function:	Quadratic fit to β-T curve

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13. ABSTRACT New research results on the following major questions are reported: Results on integrated DOD Voice and Data Networks include: analytical models for determining blocking and delay on an integrated link and numerical investigation as a function of traffic and design variables; algorithms for integrated network design were developed and programmed. The program is capable of designing networks for voice traffic, signaling and data traffic. A circuit switch model for determining switch and network transit delays for circuit connection set-up was developed. A methodology for classification of telecommunications routing algorithms was developed. Results on topological gateway placement include an algorithm and program for interconnecting packet switched networks, studies of cost/performance tradeoffs, and an application to interconnect the ARPANET and AUTODIN II. In the packet radio area models were developed to estimate network initialization as a function of number of repeaters, transmission rates of repeaters and station, and operation disciplines. Finally, cost trends for large volume packet switched data networks are derived which incorporate switching and transmission costs, satellite and terrestrial channels and local distribution.

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